Scheduling strategies to improve quality of service for heterogeneous data over resource constrained wireless mesh networks

by

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Declaration

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Abstract

Scheduling strategies to improve quality of service for heterogeneous data over resource constrained wireless mesh networks

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Cost and bandwidth plays a major role in many telemetry and Internet of Things (IoT) application network implementations. Wireless Mesh Networks (WMNs) based on single-radio single-channel (SRSC) networks will likely attract more deployments if the required quality of service (QoS) can be provided. Carrier sense multiple access with collision avoidance (CSMA/CA) is more widely used in WMNs where access to the network is decentralized and each node makes its own decision on when to access the channel. CSMA/CA in WMNs faces an increase in packet loss and contention with an increase in the number of hops to reach the destination, compared to nodes that are closer to the destination, resulting in an increase in collisions and wastage of bandwidth as the packets have to be re-transmitted. The distributed coordination function (DCF) approach does not provide data differentiated priority services, while the enhanced distributed channel access (EDCA) method was mainly designed for delay sensitive non-elastic applications to provide differentiated services. With EDCA, an unfairness problem exists, where higher priority data can starve low priority data under high load scenarios while EDCA performs poorly in multi-hop networks.

This dissertation focuses on improving QoS by reducing packet loss, reducing collisions and improving fairness to prevent starvation in low cost SRSC WMNs. This research hypothesizes that these problems can be addressed by first selecting a packet for transmission and then performing the channel contention by removing the internal contention mechanism. It also asserts that the queue selection mechanism plays a critical role in the achievable QoS. The research also hypothesizes that hybrid configured network layouts using DCF can improve performance. Hybrid and homogeneous configured network layout strategies have been investigated to support the research. Five medium access control (MAC) layer schedule before contention (SBC) mechanisms have been developed, namely adaptive weighted round robin (AWRR), roulette wheel sampling (RWS), RWS-AGE, congestion control and fairness scheduling strategies (CCFS) and queue load control priority (QLCP). The performance of these strategies is compared to EDCA and DCF networks through simulations.

It was found that the CCFS mechanism tends to starve lower priority data under heavy loads and performs poorly. RWS-AGE showed the least packet loss in homogeneous configured network layouts. A random weighted selection strategy with an age counter performs better than a weighted round robin strategy. If the lower priority data are not starved, it helps to lower packet loss as they use larger CW ranges for the back-off. An important ingredient in reducing packet loss in hybrid configured network layouts is by using DCF as it has a larger CW range and also reduces collisions. To further verify that RWS-AGE reduces packet loss, the strategy was implemented on the FIT-IoT Lab test bed and its performance was verified. A novel analytical model for the end-to-end delay for SBC strategies following Markovian theory has been developed. The results from this research support the initial hypotheses and provide important guidelines for network implementation in resource constrained WMNs carrying heterogeneous elastic traffic for a variety of applications.

Uittreksel

Skeduleringsstrategieë ten einde kwaliteit van diens te verbeter vir heterogene datatransmissie oor bandbeperkte radionetwerke

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Koste en bandwydte speel 'n belangrike rol in baie telemetrie-, asook Internet van die Dinge (IoT) netwerke. Enkelkanaal multihopradionetwerke (WMNs) sal waarskynlik wyer toegepas word as die vereiste kwaliteit van diens (QoS) verseker kan word. Kontensievermydingstrategie (CSMA/CA) word algemeen gebruik in hierdie tipe netwerk, waar elke node self beding vir toegang tot die netwerk. Waar bandwydte beperk is, soos algemeen die geval in meer landelike toepassings, gee hierdie strategie aanleiding tot aansienlike toename in pakkieverlies en gevolglike onderbenutting van bandwydte. Subnodes nader aan die eindnode, kry tipies beter diens as ander verder weg. Geprioritiseerde diens word nie ondersteun deur die verspreide koördinasiefunksie (DCF) benadering nie en die verbeterde verspreide kanaaltoegangsmetode (EDCA) is hoofsaaklik ontwerp vir stelsels sensitief vir vertraging en die voorsiening van gedifferensieerde dienste. EDCA vertoon ook 'n onregverdigheidsprobleem, waar hoër-prioriteit data die vloei van laer-prioriteit data feitlik heeltemal kan stop by hoë beladings. EDCA vertoon ook swak in multi-hop netwerke. Hierdie verhandeling fokus op die verbetering van QoS deur vermindering van pakkieverlies, die vermindering van botsings en die verbetering van regverdigheid en deurset in laekoste enkelkanaal WMNs. Hierdie navorsing toets die hipotese dat probleme verminder kan word deur aanvanklike pakkiekeuse en dan meer doeltreffende oordrag deur verwydering van die interne kontensiemeganisme.

Die proefskrif ondersoek ook dat die tou-seleksiemeganisme 'n kritieke rol speel t.o.v. die haalbare QoS. Die navorsing bepaal verder of hibriede netwerkuitlegte tesame met DCF, werkverrigting kan verbeter. Hibriede- en homogene netwerke is in hierdie proses ondersoek. Vyf meganismes vir skedulering voor toegang (SBC) tot die kommunikasielaag (MAC), is ontwikkel. Hulle is die geweegde rondomtalie- (AWRR), roulette-wiel monstermetode- (RWS), twee metodes vir vloeibeheer en regverdigheidverbetering- (CCFS en RWS-AGE) en tou-prioriteitsbeheer (QLCP). Die werkverrigting van hierdie

strategieë is uitvoerig d.m.v. simulasie vergelyk met EDCA en DCF.

Daar is bevind dat die CCFS meganisme geneig is om laer-prioriteit data ernstig te benadeel, met algemene swak werkverrigting onder hoë belading. RWS-AGE het die minste pakkieverlies getoon in homogene netwerke. 'n Ewekansige geweegde seleksiestrategie met ouderdomsteller vaar beter as 'n geweegde rondomtalie strategie. As die laer-prioriteit data nie so drasties beperk word nie, verminder dit pakkieverlies, omdat hulle ân groter kontensievenster gebruik. Die toepas van DCF is 'n belangrike aspek om pakkieverlies te verminder in hibriede netwerke, aangesien dit 'n groter kontensievenster het en ook botsings verminder. Om verder te bevestig dat RWS-AGE pakkieverlies verminder, is die strategie geïmplementeer op die Inria FIT-IOT Lab toetsbed en die werkverrigting geverifieer. Verder is 'n nuwe analitiese model, wat berus op 'n Markov benadering, ontwikkel vir die end-tot-end vertraging in enkelkanaal multihop radionetwerke. Die resultate van hierdie navorsing ondersteun die aanvanklike hipotese en verskaf belangrike riglyne vir netwerkimplementering in toepassings onderhewig aan netwerkbeperkings.

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Dedications

To my parents

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Nomenclature

List of Abbreviation

AC	Access Category
ACK	Acknowledgement
AIFS	Arbitration Interframe Space
AODV	Ad-hoc On Demand Distance Vector
APHD	Adaptive Per Hop Differentiation
APS	Adaptive Packet Scheduling
ASMC	Absorbing State Markov Chains
A-TXOP	Adaptive Transmission Opportunity
AWRR	Adaptive Weighted Round Robin
BE	Back-off Exponent
CCA	Clear Channel Assessment
CCFS	Congestion Control and Fairness Scheduling
CCK	Complementary Code Keying
CDMA	Code Division Multiple Access
CoAP	Constrained Application Protocol
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CTMS	Continuous Time Markov Chain
CW	Contention Window
DCC	Dynamic Contention Control
DCF	Distributed Coordination Function
DES	Discrete Event Simulators
DIFS	DCF Interframe Space
DNS	Domain Name Systems
DSDV	Destination Sequenced Distance Vector
DSR	Dynamic source routing
DSSS	Direct Sequence Spread Spectrum
DTMC	Discrete Time Markov Chain
EDCA	Enhanced Distributed Channel Access
EDCAF	Enhanced Distributed Channel Access Function

ETT	Expected Transmission Time
ETX	Expected Transmission Count
FDMA	Frequency Division Multiple Access
FHSS	Frequency Hopping Spread Spectrum
FIFO	First In First Out
FTP	File Transfer Protocol
HCCA	HCF Controlled Channel Access
HCF	Hybrid Coordination Function
HOL	Head of Line
HSAN	Hierarchical Stochastic Activity Networks
HWMP	Hybrid Wireless Mesh Protocol
ICT	Information and Communication Technology
INETMANET	Internet Networking MANAET
IoT	Internet of Things
ISM	Industrial, Scientific and Medical
LLC	Logical Link Control
LR-WPAN	Low Rate Wireless Personal Area Network
MAC	Medium Access Control
MANETs	Mobile Ad Hoc Networks
MIMO	Multiple-Input Multiple-Output
MPR	Multipoint Distribution Relays
MRMC	Multi-Radio Multi-Channel
MSDU	MAC service data units
NB	Number of Back-offs
OFDM	Orthogonal Frequency Division Multiplexing
OLSR	Optimised Link State Routing
OSI	Open Systems Interconnection
PCF	Point Coordination Function
PHY	Physical Layer
QLCP	Queue Load Control Priority
QoS	Quality of Service
RANN	Root Announcement
ReAP	ReAllocative Priority
RM-AODV	Radio metric Ad-hoc On Demand Distance Vector
RREP	Route Reply
RTS/CTS	Request to Send/Clear to Send
RWS	Roulette Wheel Sampling
SBB	Schedule Before Back-Off
SBC	Schedule-Before-Contention

SIFS	Short Interframe Space
SRMC	Single-Radio Multi-Channel
SRSC	Single-Radio Single-Channel
SSH	Secure Shell
TC	Topology Control
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
TDMA	Time Division Multiple Access
TFTP	Trivial File Transfer Protocol
TXOP	Transmission Opportunity
UDP	User Datagram Protocol
VCH	Virtual Collision Handler
VSAT	Very Small Aperture Terminal
WiMax	Worldwide Interoperability for Microwave Access
WLANs	Wireless Local Area Networks
WMN	Wireless Mesh Networks
WRP	Wireless routing protocol
WRR	Weighted Round Robin
WSN	Wireless Sensor Network

List of Symbols

I	Identity Matrix
0	Zero Matrix
R	Transition matrix from a non-absorbing state to an absorbing state
Q	Transition matrix from a non-absorbing state to a non-absorbing state
F	Fundamental Matrix
t	Number of steps to reach an absorbing state
B	Probability of reaching an absorbing state from a non-absorbing state
c	Column Matrix whose entries are all one
h	Denotes the h^{th} hop node in a multi-hop network
TS_h	Successful transmission time at the h^{th} hop in a multi-hop network
BO	Back-off duration
NR_h	Number of retransmissions at the h^{th} hop link
P_c	Probability Collision on the channel
P_{error}	Probability of error on the channel
CW_{min}	Minimum CW size
M	Maximum number of exponential increases allowed for the CW range
BER	Bit Error Rate
N	Number of Nodes

ρ	Traffic density or utilization factor
τ	Transmission probability
P_{idle}	Probability of the channel being idle
P_s	Probability of successful transmission on the channel
Pkt_s	Number of packets in the node
$TS_{j,h}$	Time for successful transmission on the h^{th} hop link for the j^{th} priority class
n	Priority class queue
$BO(j)$	The back-off duration for the j^{th} priority class
$NR_{j,h}$	Number of retransmission at the h^{th} hop link for the j^{th} priority class
ρ_j	Traffic density or utilization factor for the j^{th} priority class
$\overline{x}_j$	Mean of the j^{th} priority class
$\overline{x}_j^2$	Second Moment of the j^{th} priority class
S_{cw}	Contention Window range size
i	Back off stage
M	Retry limit
j	Priority class
J	Number of priority classes
w_j	Transmission probability weight of the j^{th} priority class
p_j	Queue selection probability of the j^{th} priority class
G_T	Antenna Gain of the Transmitter
G_R	Antenna Gain of the Receiver
N_t	Number of packets transmitted
N_r	Number of packets received
PW_j	Scheduling selection weights assigned to the priority class

List of Symbols with units

ACK	Time to send an acknowledgement	[s]
W_h	Access delay time at the h^{th} hop node	[s]
$W_{n,h}$	The access delay time at the h^{th} hop link for the n^{th} priority class	[s]
$AIFS(j)$	AIFS duration for the j^{th} priority class	[s]
$PropDelay$	Propagation Delay	[s]
L	Size of a packet including its header and payload	[bits]
R_t	Average transmission rate	[Mbps]
TC	Time duration in the event of a collision	[s]
D	End-to-End delay	[s]
λ	Arrival rate	[bits/s]
μ	Departure rate	[bits/s]
λ_j	Arrival rate for the j^{th} priority class	[bits/s]
μ_j	Departure rate for the j^{th} priority class	[bits/s]

P_R	Received power	[W]
P_T	Transmitted signal power	[W]
λ_{WL}	Wavelength	[m]
d	Distance between the transmitter and receiver	[m]
h_t	Height of the transmitter	[m]
h_r	Height of the receiver	[m]
y_j	Normalised throughput of the the j^{th} priority class	[Mbps]

Chapter 1

Introduction

1.1 Introduction

Wireless Mesh Networks (WMNs) have been an active area of academic research for over a decade due to their attractive characteristics. Some of these are (1) that they prevent a single point of failure as node failures do not result in the network becoming dysfunctional, as alternate routes are available immediately, (2) provide low deployment cost, (3) and provide easy implementation for the extension of existing networks [1,2]. For data packets to reach the destination from the source, there are usually more than one possible route that can be used [1].

The key challenges in application of carrier sense multiple access with collision avoidance (CSMA/CA) in WMNs for single-radio single-channel (SRSC) networks are improving quality of service (QoS) by reducing collisions, reducing packet loss [3–7] and improving intra-node fairness under heavy load scenarios [8–10]. In this dissertation, the term heavy load is used to refer to the network conditions when queuing starts and packets start to queue up in the node. Limited work has been done to address performance in single-radio single-channel (SRSC) multi-hop networks. SCRC are considered to be more promising deployment technologies in a variety of telemetry and IoT applications due to lower cost, compared to single-radio multi-channel (SRMC) and multi-radio and multi-channel (MRMC) technologies. The internet bandwidth in rural implementations is also usually very limited due to cost [11,12]. Collisions result in wastage of bandwidth as packets have to be re-transmitted. A reduction in collisions and packet loss will allow the bandwidth to be utilized more efficiently. Fair scheduling can be classified into different categories, such as hard fairness, max-min fairness, proportional fairness, mixed-biased fairness and maximum throughput [13,14]. The type of fairness studied in this investigation is to access the channel fairly between the different priority queues in a node in order to maximize throughput to prevent starvation of lower priority data, but at the same time to give higher priority data a higher probability to access the medium. The main role of a scheduling algorithm is to enable the sharing of resources and to provide QoS by choosing the next packet for transmission [15].

CSMA/CA was originally devised for peer-to-peer, single-hop wireless networks based on the IEEE 802.11 standard. In a single hop network, the destination is usually the immediate neighbour within its transmission range. In multi-hop networks, the destination can be out of the sender's transmission range and can be a few hops away. Data may need to traverse the intermediate or neighbour nodes to reach the destination. As a consequence, the end-to-end delay can become large in multi-hop networks and the collision probability increases due to contention with neighboring nodes on the same channel [3,4]. The CSMA/CA contention based strategy results in large uncertainties as to when a packet will arrive at the destination, if it arrives at all. Therefore, the primary CSMA/CA scheduling strategy performs poorly in wireless multi-hop networks with poor QoS [16].

The research presented focuses on QoS performance improvement of CSMA/CA in multi-hop SRSC networks for applications that require higher reliability and have data of different priority levels. These applications include smart grid, smart health, water utilities, gas utilities, smart agriculture and smart buildings. Data has been classified into three categories, namely high priority (HP), medium priority (MP) and low priority (LP). The design strategies proposed in this work are therefore based on these three priority traffic classes, namely high, medium and low to provide enhanced QoS service in terms of reliability and fairness. For these applications, the end-to-end delay must be within the application tolerable range which is less than 500 *ms* for high priority data and 2 s to 5 s for low priority data as highlighted in chapter 2 [17,18].

1.2 Motivation

Despite numerous implementations and use of the IEEE 802.11 standard in WMNs [19], there are still limitations that affect the performance of operation in single-radio single-channel networks. The characteristics of data transmission and communication in WMNs limit the performance [2,3,20–22].

The motivation of the work as set out in this dissertation is threefold:

1. The first motivation is that the IEEE 802.11 is already widely deployed and used in many networks. An improvement in performance of CSMA/CA when applied to single-radio single-channel wireless multi-hop networks will allow such relatively limited network capacity to be better utilised and more easily extended.

2. The second motivation is to improve performance of multi-hop with the shared medium access, carrying heterogeneous data with different priority classes. There is a general need to improve QoS in multi-hop SRSC networks.

3. The third motivation, flowing from the first two, is that there is a growing need in more rural African environments for telemetry and control for smart grid applications, water resource management and intelligent farming applications. Due to the ubiquitous availability of economical Industrial, Scientific and Medical (ISM) band based IEEE 802.11 type equipment, the implementation thereof is attractive. However, these typical wide distribution scenarios also present performance constraints due to bandwidth limitations. Enhanced throughput due to mitigation thereof by means of innovative access strategies, will increase the feasibility of implementation of such cost effective solutions to the areas concerned.

1.3 Research Problem

CSMA/CA (which is a contention based strategy) is more suitable compared to contention-free strategies such as time division multiple access (TDMA) in multi-hop WMNs as it does not require time synchronization by a central device. In multi-hop networks, a significant drop in performance is observed due to an increase in the contention for the channel resulting in an increase in collisions. Many telemetry and Internet of Things (IoT) applications consist of heterogeneous data in the network. The existing Enhanced Distributed Channel Access (EDCA) data differentiation mechanism consists of an internal contention mechanism for channel access by the different queues which tend to starve lower priority data in high load scenarios. This leads to unfairness between the different data flows [23,24]. EDCA which is originally designed for single-hop networks is also known to perform poorly in multi-hop networks [3,21,22]. The work as presented is focused on improving QoS in single-radio single-channel multi-hop networks in heavy load scenarios. The QoS issues are mainly experienced when packets start to queue within a node.

The following hypotheses are proposed:

Hypothesis 1: The internal contention mechanism in EDCA results in starvation of the lower priority data under heavy loads. Therefore, it is hypothesised that a replacement of this internal contention mechanism with a predefined deterministic weighted round robin scheduling queue selection mechanism can improve fairness and reduce packet loss in multi-hop networks.

Hypothesis 2: The operation of the scheduling queue selection mechanism has an effect on the global performance of the multi-hop network in terms of the achievable QoS. Therefore, it is hypothesised that strategies that do not starve lower priority data but give higher priority data a higher chance to access the medium can reduce packet loss and the number of collisions in WMNs compared to the EDCA contention based strategy.

Hypothesis 3: The Distributed Coordination Function (DCF) used by CSMCA/CA does not classify data or provide differentiated treatment to data of different priority levels. The contention window (CW) range values in DCF are large and it provides a high degree of fairness. Congestion in a node can result in dropped packets and hence an increase in packet loss. A load control scheduling strategy for gateway nodes subjected to more traffic load can reduce packet loss. It is hypothesised that this scheduling strategy in a hybrid configured network layout where different nodes are assigned different scheduling strategies with some of these devices assigned DCF, can result in a reduction in packet loss over homogeneous configured EDCA network layout implementations.

Hypothesis 4: The scheduling strategy as well as the network layout plays a critical role in the QoS achievable in a network. Therefore, it is hypothesised that networks with differentiated edge and core nodes, hybrid configured network layout schemes with the use of DCF can reduce packet loss as well as the number of collisions compared to their homogeneous configured network layout implementation.

1.4 Research Objectives

As stated in the introduction, the key challenges in application of CSMA/CA in WMNs is reducing collisions, reducing packet loss and improving fairness. To achieve these objectives, the objectives have been classified into primary and secondary objectives. The primary objectives perform verification through simulations. The secondary objectives are to provide an alternate mode of testing to support the primary findings.

1.4.1 Primary Objectives

1. The internal contention mechanism in a node that implements EDCA contributes to the unfairness and starvation problem. The main objective is to investigate the effect of removing this internal contention mechanism by developing novel scheduling mechanisms without internal contention mechanisms for SRSC multi-hop network scenarios. We call these approaches schedule-before-contention (SBC) strategies.

2. The EDCA and DCF strategies result in high packet loss when gateway or bottle-neck nodes experience heavy loads. The objective is to develop a novel scheduling strategy that prevents packet loss due to load level at the MAC layer at these gateway or bottle-neck nodes. The novel strategy will consider the load level for each data priority class in a node.

3. To ascertain the performances of these distributed scheduling mechanisms developed under heavy load through simulations. The baseline to which the proposed strategies will be compared is DCF in the IEEE 802.11 standard and EDCA in the IEEE 802.11e standard for data differentiated services in WMNs.

4. To investigate the performance of the load control strategy in hybrid configured network layout settings with different nodes assigned different scheduling strategies.

1.4.2 Secondary Objectives

1. To implement and test the scheduling strategy that gives the best performance in terms of packet loss on a physical test bed to verify that there is a reduction in packet loss.

2. To develop an analytical model for the schedule-before-contention scheduling approach in multi-hop scenarios to verify the simulated end-to-end delay results.

1.5 Contributions

The contributions from the investigations as carried out are as follows:

1. It has been shown via simulations that with data of different priority levels in a network, DCF performs better in multi-hop networks than EDCA in terms of less packet loss.

2. Five MAC layer scheduling strategies to improve fairness and reduce collisions have been developed. We refer to these strategies as schedule-before-contention (SBC) packet scheduling strategies as they do not have an internal contention mechanism as well as they schedule a packet for transmission before the contention for the medium takes place. The SBC strategies have different packet selection mechanisms called the adaptive weighted round robin (AWRR), the roulette wheel sampling (RWS), the RWS-AGE, the congestion control and fairness scheduling (CCFS) and the queue load control priority (QLCP) scheduling mechanism. The performance of these strategies has been compared with EDCA and DCF in multi-hop mesh network scenarios through simulations. It has been shown that the removal of the internal contention mechanism with a scheduling mechanism that does not starve lower priority data such as AWRR, RWS and RWS-AGE does improve fairness under heavy loads, as well as reduce packet loss in multi-hop networks. A packet loss reduction over EDCA of between 9.4% and 18.4% is observed with AWRR, between 9.6% and 24.5% with RWS and between 14.5% and 21.1% with RWS-AGE under high loads on average calculated.

3. The QLCP scheduling strategy has been developed for nodes that are subjected to higher load levels in a network. The performance of this strategy has been investigated in hybrid configured network layouts with different scheduling strategies assigned to edge and core routers in the network and compared to homogeneous configured EDCA and DCF network layouts through simulations. It has been shown that with the edge routers using QLCP and the core routers using DCF, the performance of the network is improved with a packet loss reduction of 21% on average calculated compared to EDCA. It has also been shown that the choice of the scheduling strategy must be dependent on the network architecture and has a significant impact on the QoS achievable in WMNs.

4. The RWS-AGE strategy is ideal for implementation in homogeneous configured network layouts with all the nodes assigned the same scheduling strategy for networks that require high reliability and can tolerate slightly higher delay than EDCA. A packet loss reduction of between 14.8% and 21.1% with RWS-AGE under heavy loads is observed, while an increase between 7.5 and 25.9 *ms* for high priority data, an increase between 14.7 and 75.8 *ms* for medium priority data and a decrease between 47.7 and 88.8 *ms* for low priority data compared to EDCA is observed. The use of TXOP has been shown to be a possible solution to reduction of packet loss and end-to-end delay further for the higher priority data classes in SBC strategies. A further packet loss reduction of 10.5% is observed with AWRR for HP data and 9% for MP and LP data under high contention and load scenarios. An end-to-end delay reduction of 20.4 *ms* is observed with AWRR for HP and MP data and up to 53.2 *ms* for LP data with RWS-AGE is observed.

5. In hybrid configured network layouts where it is possible to differentiate between edge and core nodes, a hybrid setup will give better performance. If the network requires high reliability, but can tolerate slightly more end-to-end delay, a hybrid design layout, where DCF is configured in the edge nodes and RWS-AGE is configured in edge nodes will be ideal. If the network requires low end-to-end delay and can tolerate slightly more packet loss, a hybrid design layout, where DCF is configured in the core nodes and EDCA is configured in edge nodes will be an ideal choice.

6. The RWS-AGE strategy has been implemented in the Contiki operating system on sensor nodes on the FIT-IoT Lab test bed and packet loss reduction performance improvement observed. A packet loss reduction of between 12.4% and 13.9% is observed on average calculated under heavy loads compared to the default CSMA/CA mechanism.

7. An analytical model for the end-to-end delay for schedule-before-contention has been developed and tested with the RWS and DCF strategies by comparison with simulation results for multi-hop networks. The analytical model is made up of the access delay model derived, an absorbing state Markov chain model to determine the expected number of transmissions and derived equations to calculate the expected end-to-end delay by using the values obtained from the access delay model and the expected transmission model.

8. Guidelines based on the results on the performance of these scheduling strategies have been proposed for network planning, application and network optimization in multi-hop networks.

1.6 Dissertation Overview

Figure 1.1 presents the structure of this dissertation.

Chapter 2 presents an overview of WMN architecture and components; an overview of routing protocols in WMNs; an overview of the IEEE 802.11 standard, CSMA/CA and data priority

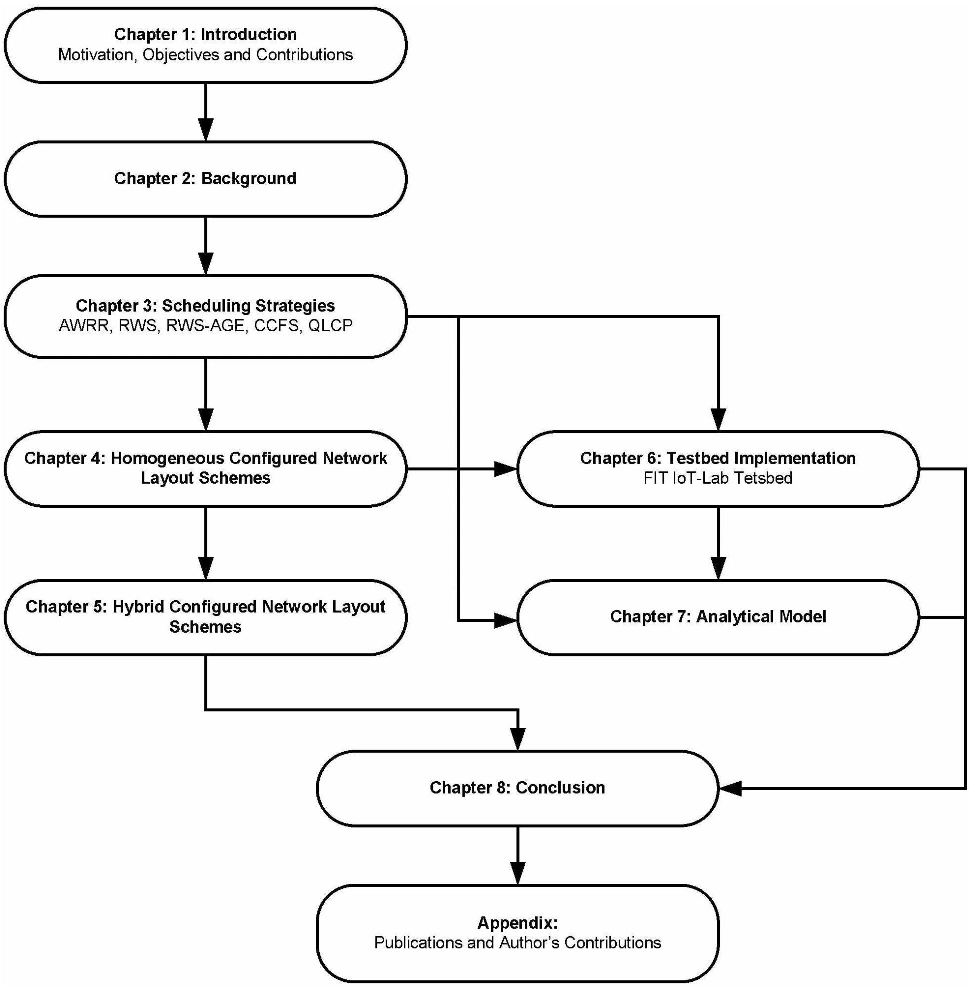


Figure 1.1: A brief structure overview of the dissertation.

services in CSMA/CA, some WMNs MAC challenges and the MAC challenges that we address in this work, and lastly an overview of literature for research conducted to addresses the fairness MAC layer challenges for heterogeneous data types. This chapter forms the foundation of the work done as presented.

Chapter 3 introduces the MAC layer scheduling strategies proposed in this research to improve QoS in multi-hop WMNs. The scheduling strategies first choose a packet for transmission from one of the priority queues and then perform the contention period to gain access to the channel for transmission. These strategies are referred to as schedule-before-contention (SBC) packet scheduling strategies. Five MAC layer scheduling mechanisms for the SBC strategies have been proposed in this chapter. The scheduling mechanisms are called the adaptive weighted round robin (AWRR), the roulette wheel sampling (RWS), the RWS-AGE, the congestion control and fairness scheduling (CCFS) and the queue load control priority (QLCP) scheduling mechanism. The strategies and packet selection procedures are explained. For the performance evaluation of the SBC scheduling mechanisms, the concept of homogeneous configured network layouts and hybrid configured network layouts has been applied. Homogeneous configured network layout schemes are the layouts in which all the nodes in the network are assigned the same scheduling strategy. Hybrid configured network layouts are layouts where different nodes are assigned different scheduling strategies.

Chapter 4 presents the an overview of the homogeneous configured network layout scheme concept, the motivation for the experiments, the simulation environment, simulation parameters and the performance metrics used to analyze the performance and to conduct a comparative analysis of the schedule-before-contention (SBC) scheduling strategies in the homogeneous configured network layouts, the results thereof and the discussion of the results.

Chapter 5 presents the hybrid configured network layouts investigations, motivation for the experiments, the experimental setup overview, the results and the discussion of the results. The best performing homogeneous configured network layout scheduling strategy was implemented on the FIT IoT-lab test bed to verify that it does indeed reduce packet loss. The RWS-AGE scheduling strategy was implemented in Contiki that utilises CSMA/CA as the default MAC layer scheduling strategy.

Chapter 6 presents an overview of the test bed implementation and the results.

Chapter 7 presents the analytical model developed to support the end-to-delay results. The analytical model computes the number of transmissions through an absorbing Markov chain model. The resulting number of transmissions feeds into the derived end-to-end delay equations, to obtain the final end-to-end delay expected.

Chapter 8 concludes the dissertation and presents possible future research directions.

Chapter 2

Background

2.1 Introduction

This chapter presents the background information for the work described in this dissertation. This chapter provides an overview of wireless mesh network (WMN) architectures, a brief overview of routing protocols in WMNs, an overview of the familiar IEEE 802.11 standards used in WMNs, CSMA/CA and data priority services in CSMA/CA, some network application requirements, some WMNs challenges and lastly a literature survey on research conducted to addresses challenges which are the main concern of this dissertation. The related work also highlights why some of the current scheduling strategies are not appropriate for SRSC resource constrained wireless mesh networks.

The seven layer open systems interconnection (OSI) layered model provides a framework for protocols operating at each layer and defines the roles to be carried out by each layer. Each layer has its own protocols which operate at the layer for which it is designed [25]. For example, in multi-hop wireless networks, the physical layer which is the lowest layer in the OSI model, is responsible for the transmission of bits between the sender and the receiver (or intermediate node) over the wireless medium. This layer also describes the electrical and mechanical properties and also the modulation technique used. The data link layer is made up of two sub-layers, (1) the logical link control (LLC) and (2) the medium access control (MAC). The MAC controls access to the channel for transmission of data, if it has data, and also performs scheduling to gain access to the channel for packet transmission. The LLC is responsible for frame synchronization, error checking and flow control. WMN architectures are presented in section 2.2. In these architectures, data is forward data through intermediate nodes in a multi-hop fashion with choice of one or more paths for data to reach the destination. The paths for routing are chosen and set up by routing protocols. Choosing the path for data transmission in multi-hop networks is known as routing. Routing is a function of the network layer in addition to network addressing. Section 2.3 presents an overview of routing protocols. The lower three layers of the OSI model, namely the network, MAC and physical layers, play a significant role in the network performance of multi-hop networks. The literature discussed in this chapter is mainly focused on these three layers.

The IEEE 802.11 standard was initially developed to provide wireless connectivity to devices in wireless local area networks (WLANs) instead of wired connectivity. This standard has been in use for over a decade. The IEEE 802.11 standard has been used in many WMNs implementations and has become the predominant standard in WMNs [19]. An overview of the IEEE

802.11 standards is presented in section 2.4. Despite the widespread use of the IEEE 802.11 in multi-hop WMNs, it has limitations that still need to be addressed. Since transmission on the medium can be heard by anyone capable of detecting the transmission, a node cannot immediately transmit on the channel without choosing a suitable period. If two nodes send data on the medium at the same time, a collision takes place. Medium access techniques are used to control access to the channel for transmission of data. These medium access control (MAC) scheduling techniques fall into two groups, namely random access contention techniques and contention-free techniques. The contention free techniques gain access to the medium through the control of a controller device. These techniques include frequency division multiple access (FDMA), time division multiple access (TDMA) and code division multiple access (CDMA). Random access techniques make their own decisions as to when it is suitable to access the medium. The carrier sense multiple access with collision avoidance (CSMA/CA) protocol, a random access technique, has been widely used in WMNs. Contention free strategies are known to provide good performance and QoS in single-hop networks through the use of a pre-defined controller. However, the situation is different in WMNs. For successful operation the popular contention free technique, namely time division multiple access (TDMA), requires perfect time synchronization between the sender, the intermediate nodes and the receiver node. This is difficult to execute in multi-hop networks as the nodes can be separated over large distances. Without synchronization, the clocks of the various nodes in the network may not have a consistent view of the global network, thus degrading performance. On the other hand, CSMA/CA does not require time synchronization and therefore is in many WMN implementations [16]. CSMCA/CA is used as the foundation MAC scheduling strategy in this dissertation work. Therefore, in section 2.5 the operation of CSMA/CA is presented and in section 2.6 the operation of CSMA/CA for data differentiated services is presented.

A variety of smart applications consist of heterogeneous type of data from different network applications. A brief overview of the requirements of these applications of priority services in these networks is highlighted in section 2.7.

Section 2.8 then presents some WMN challenges that are due to its architecture, CSMA/CA scheduling approach or due to routing over multi-link applications. Section 2.9 presents a summary of some of the current research efforts that address some of these challenges and highlights the unresolved issues for single-radio single-channel WMNs.

2.2 Wireless Mesh Network Architectures

For a WMN to exist, some devices need to be connected in a mesh layout. These mesh devices can be mesh routers, mesh gateways or mesh clients. In most implementations, the mesh is found in the backbone network through the use of mesh routers. These mesh routers are usually stationary and communicate in a multi-hop fashion. Data packets may travel over multiple links to reach the destination. The client nodes from the different network domains connecting to the mesh backbone routers can be stationary or mobile, depending on the application [2]. In smart applications and telemetry networks, the nodes are usually stationary. The mesh gateway devices provide access to the internet or connect the different network domains to the backbone network. They mainly carry traffic into and out of the mesh backbone network. In some scenarios, the mesh link can be formed between the mesh clients depending on the application or the need to extend coverage [26]. The main advantage of having mesh connectivity is that, when a link has poor quality or a node becomes dysfunctional, the network is able to dynamically use alternative routes [2]. In general routing of data in WMNs takes place in a multi-hop fashion with different paths to choose from.

Three main general network design layouts are in use for WMNs. These layout designs either use single-radio single-channel (SRSC) technologies, single-radio multi-channel (SRMC) or multi-radio multi-channel (MRMC) technologies for physical channel access. These layout designs are classified, depending on which nodes perform the mesh connectivity functionalities.

2.2.1 Backhaul WMNs

In this network layout, the mesh routers are connected as a mesh and the client nodes can connect to these mesh routers. This type of WMN network layout is very commonly deployed and can be found in many implementations [2]. An example this network layout is shown in figure 2.1.

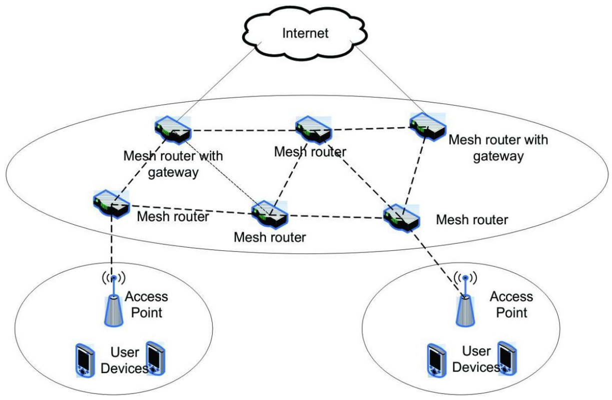


Figure 2.1: A backhaul WMN layout providing connectivity to the different network domains.

2.2.2 Client WMNs

In this network layout, the mesh connectivity is formed between the client nodes. In order to maintain mesh connectivity, these client nodes perform routing in addition to providing user access to the network [2]. An example of this network layout is shown in figure 2.2.

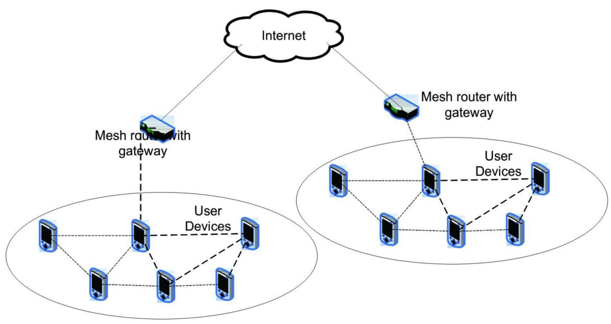


Figure 2.2: A network layout with mesh connectivity between the clients.

2.2.3 Hybrid WMNs

In this network layout, the mesh connectivity can be found in both the backbone nodes and the client nodes. The mesh client devices can gain access to the network either through a mesh router or directly [2]. Figure 2.3 presents an example of a hybrid WMN layout.

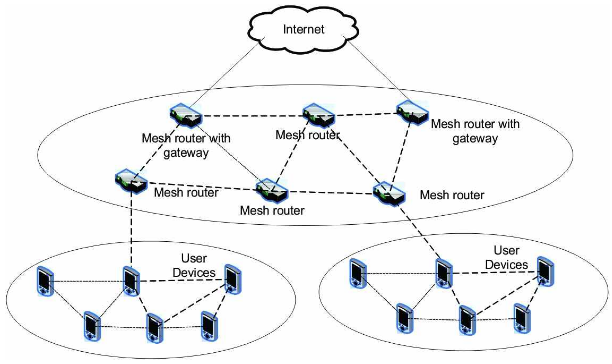


Figure 2.3: A hybrid wireless mesh network layout.

2.3 Wireless Mesh Network Routing Protocols

The WMN layouts presented in section 2.2 show that more than one path can exist to send data from a source to a destination. The choice of the route used is decided by the routing protocol. Routing protocols select a routing path and also allow data to be transmitted in a multi-hop fashion to reach the destination. A common classification approach to categorise routing protocols has been to group them into proactive, reactive or hybrid routing protocols. The proactive routing protocol groups periodically check routes to all nodes and store this information in their routing tables. Up-to-date information is distributed in the network by each node periodically distributing its routing table throughout the network. Examples of such protocols are the Destination Sequenced Distance Vector (DSDV) routing protocol and the Optimised Link State Routing (OLSR) protocol. In this investigation we have used the OLSR protocol as it is the most widely used routing protocol. The OLSR performs a periodic updates on the links in the network. This allows information on the most current state to be available in the routing tables of each node. This helps if a node failure took place in the network [27]. This protocol uses Hello and Topology Control (TC) messages to discover link changes or current link states and to spread this information throughout the network. Each node uses this information to determine a suitable next hop, using the shortest hop forwarding path for sending its data. In OLSR, Hello messages are used by the nodes to determine the 2-hop neighbour information, as well as to perform a distributed election of a set of multipoint distribution relays (MPRs). Each node in the network then selects its MPR. Nodes that are not MPRs can only read and process the routing information, but cannot perform re-transmission of these broadcast messages. In OLSR, each node also maintains a set of neighbours known as the MPR selectors. These MPR nodes can source and forward TC messages between the MPR selectors [28].

With the reactive routing protocols, a path from the source to the destination is only discovered when a node has data to send. If the source node does not have route information available for that destination in its routing table, it performs a route discovery process by sending Route Request (RREQ) messages to all the neighbouring nodes. The nodes that receive these RREQ messages rebroadcast these messages, provided that none of them is the destination node. Only the destination node will not forward these RREQ messages. When the destination node receives these RREQ messages, it responds by sending a Route Reply (RREP) message back to the node that initially sent the RREQ message. In the case that a link becomes faulty or is removed from the network, an error message is transmitted. Examples of such protocols are the Dynamic Source Routing (DSR) protocol and the Ad-hoc On Demand Distance Vector (AODV) protocol. Reactive routing protocols inject less overhead in the network, compared to proactive routing protocols such as the OLSR[29]. Hybrid routing protocols share features of both reactive and proactive protocols. The different network domains either use a reactive or a proactive approach. Domain here means sections of the network that have mobile nodes and sections of a network that have static nodes. An example of a hybrid protocol is the Hybrid Wireless Mesh Protocol (HWMP), which has mainly been developed for WMNs and is implemented in the IEEE 802.11s standard for WMNs. The HWMP protocol is designed to use reactive routing for mobile devices and proactive tree-based routing for fixed devices in the network [30]. HWMP caches each alternate route, so that if a node fails, an alternate route is available immediately [30]. The use of OLSR in this work allows the testing of the proposed scheduling strategies under the operation of a protocol which introduces overhead into the network even if the nodes are static.

2.4 IEEE 802.11 Standards

The IEEE 802.11 standard defines the lower two layers of the OSI reference model, namely medium access control (MAC) and physical layer (PHY)[31]. The features for the different IEEE 802.11 standards are summarized in table 2.1. The IEEE 802.11n standard is designed for high throughput using multiple-input multiple-output (MIMO) technology. This is achieved by using multiple antennas, multi-channels and spatial multiplexing [32,33]. MIMO increases hardware cost. IEEE 802.11ac is a newer standard which is designed to provide very high data rates up to 7 Gbps using the 5 GHz band and also uses MIMO technology as in the IEEE 802.11n standard[34]. The modulation techniques used in these standards are frequency hopping spread spectrum (FHSS), direct sequence spread spectrum (DSSS), orthogonal frequency division multiplexing (OFDM) coding and complementary code keying (CCK). All these IEEE 802.11 mentioned standards have been developed mainly for single-hop networks where the destination is within the range of the source. Although these standards can be used in multi-hop networks, they have not been optimized for these applications. For them to be used in multi-hop applications, a routing protocol is required as well.

The IEEE 802.11e standard is implemented on top of the existing standards in table 2.1 for data differentiation and is explained in section 2.6. The IEEE 802.11s standard has been developed for WMNs. This standard is an amendment of IEEE 802.11 for mesh networking which adapts either one of the IEEE 802.11 standards given in table 2.1 to provide channel access, scheduling and data transmissions. However, a routing protocol for the choice of the transmission path needs to be implemented. Although this standard uses the default HWMP protocol, an alternative routing protocol can also be used. The main feature of HWMP is that routing is implemented at the link layer rather than at the network layer.

The primary MAC scheduling technique dealt with in all the IEEE 802.11 family of standards is the distributed coordination function (DCF), which basically uses CSMA/CA. The next section presents an overview of CSMA/CA.

Table 2.1: IEEE 802.11 standards

	Legacy IEEE 802.11	IEEE 802.11a	IEEE 802.11b	IEEE 802.11g	IEEE 802.11n	IEEE 802.11ac
Frequency Band (GHz)	2.4	5	2.4	2.4	2.4, 5	5
Maximum Data Rate (Mbps)	2 Mbps	54Mbps	11Mbps	54Mbps	600Mbps	6.93Gbps
Modulation	FHSS, DSSS	OFDM	DSSS, CCK	DSSS, CCK, OFDM	DSSS, CCK, OFDM	OFDM
No. of Non-overlapping Channels		23	3	3	26	Many

2.5 Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA)

The IEEE802.11 CSMA/CA technique has gained widespread popularity in multi-hop WMNs, providing best effort services at the MAC layer, rather than guaranteed service. The MAC layer has two access mechanisms, namely the distributed coordination function (DCF) and point co-ordination function (PCF) [34]. The latter is a contention free strategy, as access to the medium is granted with the help of a controller. With DCF a node contends for the medium if it has data for transmission. In smart applications, the nodes do not gain access to the network through a central communication device (controller) such as an access point, using PCF. PCF requires all the devices to be synchronized with the central coordinator. Therefore, in these applications, the contention based, distributed DCF mechanism is more suitable. Access to the network is decentralized since each node makes its own decision on when to access the medium. CSMA/CA uses the DCF which first listens to the medium before making a decision on transmission. If a node has data to transmit, it first senses the medium for a DCF Interframe Space (DIFS) period. If the medium is found to be free of any communication for this DIFS period, the station then performs back-off by generating a random number in the range of 0 and contention window (CW) size and then counting down for this number of time slots. Initially the CW value is set to CW_{min}. This value of CW is exponentially increased every time a collision takes place. The value of CW can only take a maximum value of CW_{max}. After the re-transmission limit is reached, the packet gets dropped. The countdown time freezes every time the channel is detected to be busy and continues after the channel is free for a DIFS period. After a packet is successfully transmitted to the receiver, the value of CW is then reset to CW_{min}. The default CW_{min} value is equal to 31 and CW_{max} is equal to 1023. The cost function of the CW window is given in equation 2.5.1.

$$S_{cw} = \begin{cases} 2^i CW_{min}, & 0 \leq i \leq m \\ 2^m CW_{min}, & m < i \leq M \end{cases} \tag{2.5.1}$$

where

i represents the back off stage which is between 0 and m
M is the retry limit
S_{cw} is the contention window size

2.6 Data Priority Based Services in CSMA/CA

With DCF, data of different priority are treated equally and the data can access the channel in a first in first out (FIFO) method from the transmission queue. Differentiated services are not provided in DCF and therefore the end-to-end delay is dependent on the order in which the different priority data are received in the queue. Many network applications require differentiated services, as they generate heterogeneous data with different priority levels from the different network domains.

The IEEE 802.11e standard was developed for multimedia traffic to provide differentiated services to traffic with different QoS requirements[34] and uses the hybrid coordination function (HCF) which is also based on two mechanisms. One is a centrally-controlled mechanism which requires a controller, known as HCF Controlled Channel Access (HCCA), and the other is a contention based medium access mechanism, known as Enhanced Distributed Channel Access (EDCA) [36], not requiring the help of a controller. In this dissertation we focus on EDCA which is an enhancement of DCF and does not require a central controller.

EDCA classifies data traffic into different classes, called access categories (AC), for different traffic types and consists of up to four ACs. Each AC has specific parameters associated with its priority class for the channel contention period. The parameters are designed such that the ACs with higher probabilities have a better chance of gaining access to the medium than the lower priority ACs [37]. Data is first classified at the MAC layer so that it gets placed in one of the corresponding AC queues. The four ACs in EDCA are: background, best-effort, video and voice. EDCA modifies DIFS by introducing a new interframe spacing called Arbitration IFS (AIFS). AIFS, like DIFS, is the least time period that the medium must be sensed as being free (with no activity) before an attempt is made to transmit. After this AIFS period, a back-off is performed. The duration of the back-off period depends on the number chosen within the CW. The higher priority data have smaller CW ranges compared to the lower priority data. The minimum and maximum CW values assigned are variable, depending on the AC and are not fixed as it is in DCF [36]. Therefore, EDCA statistically provides QoS by differentiating channel access among traffic having different priority levels. For each of the ACs, the contention parameters are presented in table 2.2, namely, AIFS number (AIFSN), CW values and Transmission Opportunity (TXOP limit) values. The TXOP limit is the duration that a node can send data on the channel without having to contend for the medium. A node can send out multiple data packets one after the other from the same priority queue until the time period of transmission reaches the specific TXOP limit [3,38]. The assigned default TXOP limit values according to the standard are based on the slot time and packet size for the different data priority classes. The slot time is 20 μs in the IEEE 802.11b standard and 9 μs in the IEEE 802.11g standard [39]. Figures 2.4 and 2.5 present the scheduling mechanism of EDCA. If any AC queue has data for transmission, the node schedules transmission by each AC queue by sensing the medium to be idle for an AIFS period and then performs the back-off period. Each queue behaves as a virtual node performing sensing of the medium for the AIFS period and then counting down for the back-off duration, concurrently for each queue with data. Each node can have one or more AC queues depending on the application. The duration of the AIFS period is calculated using equation 2.6.1.

$$AIFS[N] = AIFSN[AC] * SlotTime + SIFS \tag{2.6.1}$$

where AIFSN[AC] is the number of slots.

Fig 2.5 shows the flow chart operation of EDCA. With EDCA, both internal and external collisions are experienced as explained earlier.

Table 2.2: AC dependent parameter values in EDCA

AC	Type of Data	AIFSN	Minimum CW value	Maximum CW value	TXOP limit in the IEEE 802.11a/g standard	TXOP limit in the 802.11b standard
AC [3] - Lowest Priority	Background	7	31	1023	0	0
AC [2]	Best Effort	3	31	1023	0	0
AC [1]	Video	2	15	31	3.008 ms	6.016 ms
AC [0] - Highest Priority	Voice	2	7	15	1.504 ms	3.264 ms

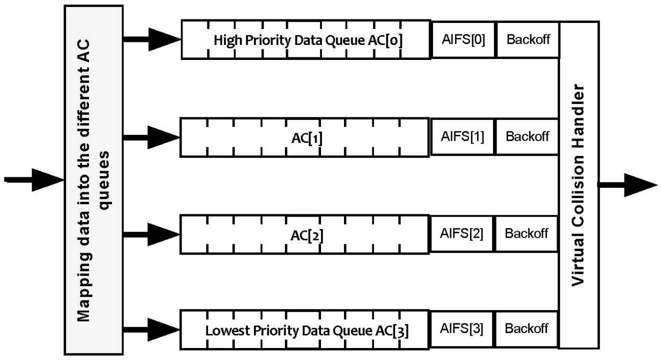


Figure 2.4: Scheduling in EDCA for the different AC category data..

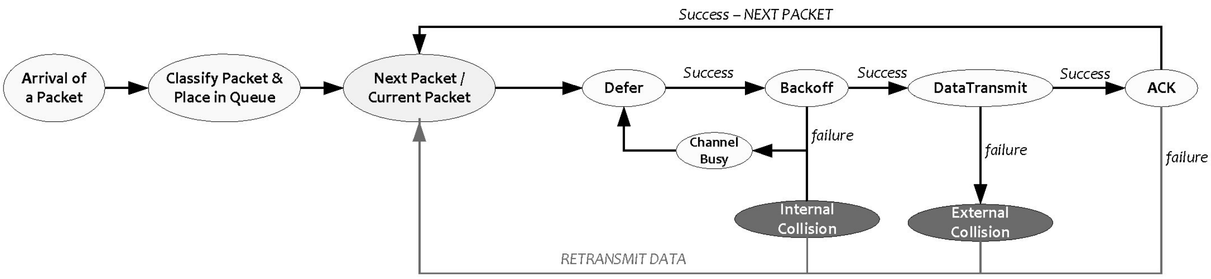


Figure 2.5: Medium access flowchart for the EDCA.

2.7 Telemetry and IoT Application Requirements

As stated in the motivation for this work, there is a growing need in rural African environments for telemetry and control networks. Rural areas mainly have underdeveloped infrastructures for services such as health and education [40]. This section presents a brief overview of some possible application areas of the proposed scheduling strategies as covered in this work.

Applications can be divided into real-time and non-real time applications. Real time services include voice and video communication, two-way telemetry and telnet [41]. According to [41], the preferred acceptable tolerable level of delay is up to 150 *ms*, while 400 *ms* is also acceptable. Speech delay up at 150 *ms* is acceptable by most users [42]. Two-way telemetry and telnet applications have a delay requirement to be kept below 250 *ms* [41]. Real-time video streaming has a maximum tolerable delay requirement of 10 seconds [41]. Non-real time services include web-browsing, file transfer protocol (ftp), high priority transactions such as ecommerce and emails. These services can tolerate delays in the range of 2 to 4 seconds [41,42].

In this section, six different IoT smart application domain areas are briefly highlighted as shown in figure 2.6. These application domains are smart grid which includes energy; smart transport which includes transportation, traffic and parking; smart education which include networks for educational use; smart health; smart farming which include both horticulture and livestock farming; and smart buildings. Smart operations are usually made possible in networks through the use of intelligent sensors and actuators; two-way communications; control and monitoring mechanisms; information and communication technology (ICT); and the internet. Each of these application environments consists of data of different priority levels. We have classified the data into three categories, i.e. high, medium and low priority. The design strategies we investigated in this work are therefore, based on these three data priority classes (high, medium and low) to provide improved QoS service in terms of reliability and fairness. Table 2.3 gives a summarised classification of priority data for these different application domains as above mentioned. Table 2.4 presents the requirements of a smart grid communication network in more detail. The advanced smart metering infrastructure can tolerate more delay than network data from fault detection networks. Detailed smart grid performance requirements in terms of latency and reliability for different smart grid applications are also stated in [17] and [18].

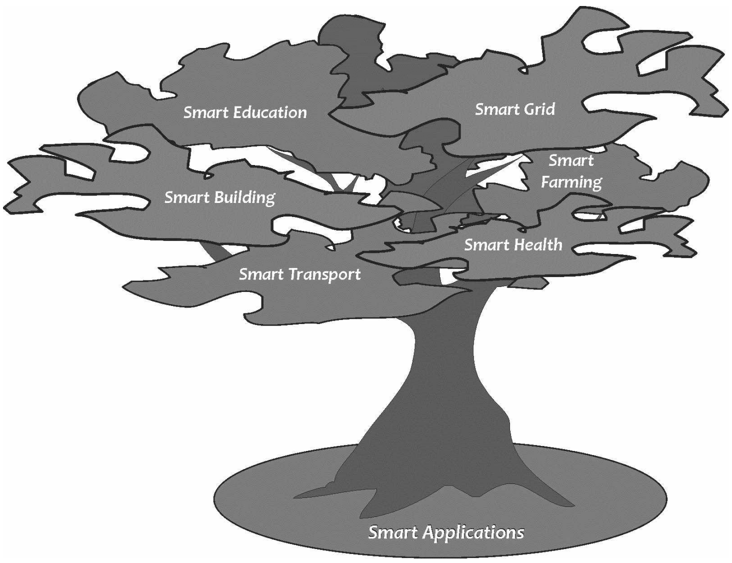


Figure 2.6: Some IoT smart application domains.

WMNs are known to provide low deployment cost. There is a growing need to improve performance of wireless multi-hop networks using CSMA/CA for priority based services in resource constrained networks for implementation for applications mentioned in this section. The resource constrained networks investigated in this work are based on SRSC.

2.8 Wireless Mesh Network Challenges

WMNs carry out transmission in a multi-hop fashion if the destination node is a few hops away. Many factors play a critical role in the performance of WMNs. Some of these factors include the network layout design, network topology, amount of traffic load, number of hops as well as number of devices in a collision domain, size of the network, the number of channels used, bandwidth capacity, coverage area and link quality for routing paths [2]. This section presents a brief overview of some of the challenges in SRSC WMNs.

Table 2.3: Data priority classification for different smart application domains.

Application	High Priority Data	Medium Priority Data	Low Priority Data
Smart Grid	Emergency Response	Automated Demand Response (ADR)	Advanced Metering Infrastructure (AMI)
	Fault Detection	Transformer Monitoring	Remote Connect/Disconnect
	Supervisory control and data acquisition (SCADA)	Direct Load Control	Voltage and Current Monitoring
Smart Education	Online tests	Audio Conferencing	Web Browsing
	Exams		Emails
	Video Conferencing		Online Libraries
Smart Buildings	Air-conditioning (HVAC) systems	Access control systems	Web Browsing
	Video Surveillance, Safety Alarms		Internet Access
	Fire protection systems		Smart Lighting Designs
Smart Transport	Ticketing	Digital Signage	Sensor Object Detection for Parking
	Payments	Transport Logistic	
Smart Agriculture and Animal Farming	Renewable energy sources	Tracking of livestock	Sensor readings such as temperature, feed level, soil moisture, Access to stock level for suppliers
Smart Health	Tele-monitoring â remote health monitoring of patients	Mobile Assistance	Office based applications, medicine use intake by patient, messages to patients

Table 2.4: Smart grid communication requirements.

Priority Category	End-to-end Delay	Reliability
High	<500ms	99-99.9%
Medium	500ms - 2s	99-99.9%
Low	2s - 5s	99-99.9%

2.8.1 Scalability

A common problem in WMNs is that of scalability. The performance degrades with an increase in the number of nodes as well as the distance (number of hops) to reach the destination node [2,43,44]. With an increase in the number of nodes, the nodes have to wait longer to gain access to the channel as the contention increase result in a drop in network capacity by an increase in collisions and an increase in packet loss [3–6]. If the destination node is a few hops away, the data may need to be transmitted over multi-hops to reach the destination. This causes more contention to access the medium compared to data that does not have to span long distances [45]. Due to the need to span the network over long distances, the need for multi-hop communication increases.

If the channel bandwidth is not efficiently utilized, it also results in a drop in performance. In [46] a study was done to change the initial contention window (CW) size depending on the number of nodes in the transmission range and the traffic load. By default this value stays the same in the IEEE 802.11 standard except the increases in the CW that takes place when a collision occurs. The CW value was made proportional to the number of nodes that try to transmit on the channel and traffic load for single-hop networks which showed performance improvement. Smaller CW range values increase the chance of collisions [47]. In EDCA, the higher priority data classes use smaller contention window sizes. EDCA is shown to perform poorly in multi-hop networks [21,22].

2.8.2 Topology

Many of the routing protocols initially proposed and developed for Mobile Ad Hoc Networks (MANETs) have been extended and adapted for use in WMNs. WMNs have some major differences compared to MANETs. The nodes in WMNs are usually static as in the case for backhaul WMNs, while in MANETs, the nodes are usually mobile. In MANETs, the prime objective has been to maintain connectivity, while for WMNs, the main objective has been to finding and using links that will give the highest performance. The routing protocols that have been designed for MANETs, therefore do not perform well in WMNs [26].

2.8.3 Interference and Channel Capacity

To lower the cost, SRSC will likely attract more implementations. Interference, link conditions and channel capacity play a major role in SRSC strategies. The performance of a WMN is greatly affected by the link conditions where the environmental conditions play a significant role [48]. The wireless link conditions can vary with time depending on the environmental conditions. The wireless link quality is subjected to channel fading, path loss, interference and shadow fading. A routing protocol may consider link load, link quality and other factors to determine the suitable path for data transmission. This is achievable by using information from other layers such as the physical layer on link quality for routing decisions. The layered design does not use information from other layers for routing [49,50].

Interference plays a significant role in the performance in multi-hop networks as shown in [51–53] and [19]. The carrier sensing range is always greater than the transmission range in cases where SRSC networks are used [52]. A problem known as adjacent channel interference (ACI) exists in these networks where "bleeding over" takes place. This causes sensing from outside of its transmission range [53]. If a node within the interference gains access to the channel and transmits data, all other nodes within the interference range have to wait until the transmission finishes, affecting the achievable capacity of the network [19,53]. WMN nodes can be designed and implemented with multiple-radio multi-channel technology which greatly increases the cost compared to SRSC. The use of multiple non-overlapping channels has been shown to increase the overall throughput and reduce inter-node interference as they allow concurrent transmission in non-overlapping channels [54].

Studies done in [55–58] have shown that the link capacity in SRSC networks drops with an increase in the number of nodes or number of hops in multi-hop networks. In [55], the authors show that the throughput reduces by $O(\frac{1}{N})$, where n is the total number of nodes. In [56] and in [58] it has been shown that for a purely ad-hoc network with a random selection of source and destination, then the capacity reduces by $O(\frac{1}{\sqrt{N \log N}})$. For chain topologies, the capacity reduces by $O(\frac{1}{N})$, where N is the total number of nodes provided that only one node can transmit at a time [57]. The main challenge that exists in SRSC networks is using the channel capacity effectively as the capacity reduces with an increase in network size.

2.8.4 Fairness and Starvation problem

The basic structure and operation of EDCA was explained in section 2.6. EDCA has been used with the IEEE 802.11s standard in WMNs. Every node that implements the default EDCA has multiple queues. If any queue has data, data is scheduled after sensing the medium to be idle for an AIFS period and a back-off period depending on the priority class. As mentioned earlier, the AC queues behave as virtual nodes and contend for the medium. In the situation when two ACs finish the contention period at the same time and try to access the medium, an internal collision takes place. This internal collision is governed by the management mechanism called the virtual collision handler. This virtual collision handler allocates channel access to the higher priority data, while the lower priority data is treated as if a real collision on the medium took place. This lower priority data contends for the medium again by exponentially increasing its CW range as per rules stated in section 2.5 [59,60]. If a node transmits successfully, it sets its CW to the initial value giving other packets in the same queue an even higher chance to be transmitted [8]. Therefore, in default EDCA strategy an unfairness problem exists where the higher priority data can starve the lower priority data [8–10].

2.8.5 Congestion

WMNs share network nodes with other users to send data as the network is built such that communication takes place in a multi-hop fashion. In multi-domain networks, the gateway devices are subjected to more load levels as they pass data from the different domains to the backbone mesh in backhaul WMNs. For traffic to be sent to the internet, it must first be routed to the gateway node that connects to the internet. These gateway devices in WMNs therefore, experience high load which results in congestion at these nodes [61]. If the capacity of the node is full, incoming packets are dropped. Also if some priority queues are becoming full, the lower priority data can be starved. When a node experiences congestion, traffic should be re-routed through less congested nodes or a MAC strategy should be in place to prevent packets from being dropped or prevent starvation to improve performance.

Many telemetry and Internet of Things (IoT) applications such as smart grid, home-automation, health-care monitoring are characterized as consisting of heterogeneous data in the network (Section 2.7). These heterogeneous data have different priority levels depending on the applications. EDCA was mainly proposed for networks carrying multimedia traffic such as voice and video. Multimedia traffic can tolerate small amounts of packet loss but require less end-to-end delay [62]. For these applications, the end-to-end delay QoS is given a high importance. WMN applications can be divided into two classes, namely delay sensitive (non-elastic) and non-delay sensitive (elastic). However, there are many non-delay sensitive applications that require a high degree of reliability (less packet loss) over delay. This is to say that they can tolerate slightly more delay provided the end-to-end delay is within tolerable ranges. Examples of these applications are smart grid, smart buildings, smart farming and smart health [63][64]. These applications carry heterogeneous type of data having different priority levels running on the same communication network. We have classified the requirements of these applications into three categories, namely high, medium and low priority. For EDCA to be used in these applications to carry data of different priority levels, it will have to be able to provide a high degree of reliability as well as provide end-to-end delay within tolerable ranges.

This section has highlighted some major challenges that exist in WMNs using SRSC. These challenges include the scalability issues; the starvation problems in scheduling; routing not considering link loads and other link conditions to prevent congestion and using poor links for routing; and congestion.

2.9 Related Work

This section presents a summary of some of the research efforts that address some of the challenges presented in section 2.8 and highlights the unresolved issues for single-radio single-channel WMNs.

The problems mentioned in section 2.8 are not new. Numerous studies exist in literature to address the routing issues in WMNs for packet loss and collisions by considering channel and load conditions. To reduce the number of collisions, or the extent of packet loss in WMNs, some studies have focused on developing routing metrics that choose routing paths with better link qualities in terms of less congestion or less interference for data to travel from the source to the destination. In WMNs, the environmental conditions vary with time and therefore, affect the wireless link condition. Cross-layer strategies that use information from adjacent or non-adjacent layer have been developed to address packet loss and fairness problems. The Expected Transmission Count (ETX) and Expected Transmission Time (ETT) routing metrics have been designed for single radio and single channel networks which consider packet loss in a path [20,26]. Routing techniques as in [65–68] have been developed for single-path routing, considering link conditions while multi-path techniques that mainly use soft computing techniques have been developed in [69–83]. The disadvantage is that these techniques either introduce more overhead on the medium and network or require more buffer memory to store additional information in their routing tables. This is not suitable for resource constrained smart applications as memory is a critical factor in hardware due to the need for low cost hardware. A survey of numerous other routing techniques proposed with novel routing metrics considering link conditions and other metrics can be found in [84]. Multi-radio and multi-channel (MRMC) techniques are also widely used to improve performance by allowing concurrent transmissions on different channels [85,86].

MRMC techniques increase deployment costs as each node needs to be equipped with more than one radio. Single-radio multi-channel (SRMC) techniques have also been seen as a technique to reduce the interference between flows and thus improving the overall performance. However, with the use of a single radio, a device can either transmit or receive (half-duplex) at a given time. If the nodes might be in the transmission range for communication, they cannot communicate successfully unless they are both configured to the same channel. Again, if there is overlapping of assigned channels with different frequencies, the performance is not necessary improved if there is interference. To address this, numerous studies exist for SMRC for channel assignment to prevent interference [87]. In SRMC, for a device to communicate over neighboring domains with different channels assigned, the transceiver needs to switch from one channel to the other which introduces delay [88]. For successful allocation of non-overlapping channels to neighboring links, a central controller is required, or either a technique has to be used that introduces more overhead in the network. Most routing protocols have been designed for single-channel techniques and thus, multi-channels may lead to inefficient routing paths in WMNs [89,90]. The channel assignment, switching and routing issues have been studied in literature for SRMC. Very little research has been done on SRSC techniques to improve performance in WMNs.

Approaches to deal with the starvation problem in IEEE 802.11e EDCA contention based single-hop WLANs has been addressed extensively by many researchers in the literature as in [91–93]. Multi-hop networks are subjected to more contention and collisions compared to single-hop networks and therefore the performance is affected considerably. Limited work has been done to address the intra-node fairness and collision increase problems in multi-hop WMNs. Since multi-hop networks have to transmit data over multiple hops to reach the destination, they are usually subjected to more contention and collisions in comparison to single-hop networks. In EDCA, differentiated services are provided by assigning different parameter values such as CW_{min}, CW_{max}, AIFS and TXOP to the different priority queues. Different studies have proposed solutions focused on varying these parameters to address the starvation and collision problems in multi-hop networks. In [3], the proposed strategy called adaptive-TXOP (A-TXOP) focuses on dynamically changing the TXOPlimit values. This TXOP interval is adjusted based on the number of packets in the queue so that video data frames that are fragmented get sent in the same TXOP period. This has shown a reduction in delay to transmit large video frames. This strategy is mainly developed for applications with large packets such as video that are broken down.

The intra-node fairness has also been investigated by changing the priority of the messages and not keeping it constant as in [3,94–97]. In [3], a dynamic ReAllocative Priority (ReAP) strategy is proposed, where the packet priorities change according to network conditions by using hop count information which is obtained from routing table of the routing protocol at the network layer. This technique has shown to improve the packet delivery ratio by 28% as compared to EDCA under heavy load. An adaptive per hop differentiation (APHD) scheme is proposed in [94] which aims at achieving end-to-end delay application requirements in multi-hop wireless networks. It does this by adjusting the data packet's priority levels. APHD is a cross-layer technique which modifies the header of the packet to carry end-to-end delay information of the application, performs node state monitoring and also adjusts the priority of the messages, based on the delay requirements. Another strategy called dynamic contention control (DCC) is proposed in [95] wherein the packet priority is dynamically adjusted according to the calculated delay per hop and the back-off timer. This strategy is mainly designed to control congestion. The mobile nodes estimate delay by using the received acknowledgement frames that are sent out every time a successful transmission occur. The priorities in [96] are made dynamic depending on the network conditions and required QoS. The problem with these techniques is that for their successful operation, they require the use of some information from the network and other layers such as load level, numbers of hops left or acknowledgments. The other problem is that the priority of a packet keeps changing across the network and extra header fields in the packet are required to keep information on the priority of the packet, end-to-end delay information and therefore introduce extra overhead into the network. These techniques are also mainly developed for multimedia applications where end-to-end delay is very critical. Non-delay sensitive (elastic) applications such as smart applications presented in section 2.7 require a high degree of reliability (less packet loss) over delay.

The starvation problem has been addressed for intra-class data in [98] through a proposed adaptive packet scheduling (APS) layout that adjusts the packet size. This strategy schedules packets from different queues by allocating resources among different service classes to achieve inter-class fairness. Introduction of separate extra queues has been studied in [99] and the use of weighted queues to address the unfairness problem was investigated in [9], [91] and [92]. The techniques in [9], [91] and [92] have been developed and tested mainly for single hop WLANs. The queues for transmission are selected in a round robin fashion according to the weights assigned. In [99] a network layer solution is proposed to address the fairness problem by having queues in the network layer and at the MAC layer. The performance of using different number of queues in the network and MAC layer was analysed while assigning different weights to the queues to gain access to the different levels of bandwidth. Another problem is that the forwarding data and originating data share the same queue. A solution is investigated of having two separate queues, one for the forwarding data and the other for the originating data and then serving them in a round robin manner. A fair queuing scheduling strategy is proposed in [93], called FQ-EDCA, whereby packets are placed in specific queues depending on the source. In this technique the original EDCA queues are replaced by a further queuing mechanism depending on the type of data. The congestion is controlled in FQ-EDCA. The disadvantage of FQ-EDCA is that it introduces more queues which require more buffer memory as well as further classifications techniques, compared to the original EDCA. FQ-EDCA requires more buffer memory.

To improve channel utilization, in [100] a scheme is proposed to reduce the number of ACKS. The MAC-layer sends acknowledgement frame if it receives a data frame successfully. Thus, this MAC-layer ACK mechanism can guarantee hop-by-hop reliability. Both IEEE802.11 MAC and TCP have ACK mechanisms. This results in extra overhead. TCK ACKs are combined with MAC layer ACKs in this approach to reduce the overhead on the channel. For video streaming applications, a study was done in [101] to improve the performance of IEEE 802.11 multi-hop wireless mesh networks to optimise throughput, minimizes packet loss and improve end-to-end delays for UDP protocol. A technique to improve the unreliable and congestion prone transport protocol namely UDP for delay sensitive applications was proposed. MAC layer information for rate control approaches for rate adaptation using Automatic Rate Fallback (ARF), Receiver Based Auto Rate (RBAR) and Frame Error Rate (FER) were used. In this strategy, the sender uses the physical layer information such as SNR, Received Signal Strength (RSS) to select an appropriate rate for transmission. The transmitter uses the RSS as the Channel State Indicator (CSI). For the successful operation of this technique, information from other layers is required.

To address congestion, in [102] a congestion contention control scheme is proposed to maximise the network throughput by exchanging prices between the source and the different network links based on the network link quality. A distributed and scalable algorithm is developed for optimal utilisation of system resources to provide end-to-end QoS. The MAC layer is optimised to improve channel access probability by reducing channel collision by using CSMA/CA with RTS/CTS. The exchange of prices introduces more overhead on the network.

For low cost resource constrained telemetry networks, strategies that do not introduce more overhead in the network and that reduce collisions will be more suitable. A reduction in collisions will assist in utilising the available bandwidth more efficiently. The issue of improving the performance of EDCA in terms of reducing packet loss, reducing collisions and improving intra-node fairness under heavy loads in resource constrained SRSC multi-hop networks for non-multimedia applications has still not been addressed. Minimum modifications to the existing EDCA strategy will allow its use to be extended to multi-hop networks.

In [9], the authors proposed a schedule before back-off (SBB) policy whereby frames are first selected for transmission, thereafter the channel access functionalities are performed. They investigated the round robin and weighted round robin (WRR) strategies, followed by variable back-off parameters in single-hop networks. Scheduling between the different priority classes is considered first, thereafter the contention periods are performed to gain access to the medium. In the work set out in this dissertation, we use a similar concept only for the channel access technique by first scheduling a packet and then performing a back-off period. We apply our study to multi-hop networks and investigate the performance with different scheduling mechanisms for the packet selection including the using of weighted round robin as the selection mechanism.

Table 2.5 summarizes the proposed MAC scheduling strategies developed for multi-queue data in order to improve performance by reducing packet loss, starvation and collisions. The table gives the main application for which the strategy was proposed, the complexity and indicates whether it was developed for multi-hop networks. The modifications to the existing EDCA are represented in terms of complexity with major modifications being categorized as high and not so complex modifications as low.

2.10 Conclusion

This chapter has presented an overview of the three main network layouts in WMNs, namely backhaul, client and hybrid WMNs. A basic overview of CSMA/CA which operates at the MAC layer and routing protocols in WMNs have been presented, followed by an overview of the IEEE 802.11 standards used in WMNs. The application requirements of some networks are discussed, followed by challenges that exist for WMNs using SRSC for resource constrained networks. Lastly, a summary of the important work done to address some of these problems in SRSC multi-hop networks has been presented. For data to be transmitted from source to destination, it may need to traverse multiple nodes or links and have various paths to choose from for routing. Extensive research has been reported, based on improving routing to choose less congested routes or routes with better link quality. Although EDCA has been in use for more than a decade and is still an active area of research, limited recent research has been found to address the unsolved issues in SRSC multi-hop networks. Only limited work has clearly been carried out to address the scalability, starvation and congestion control problems as part of scheduling strategy for SRSC multi-hop networks using CSMA/CA. The current approaches have been mainly designed for multimedia applications. They are not suitable for SRSC resource constrained settings as they introduce more overhead in the network, require more buffer memory or require information from other layers or the network. Although MRMC provides significant performance improvement, it is known to be expensive. The SRSC approaches are used in this research project.

Table 2.5: MAC Layer approaches to improving QoS for multiple queue data (classified data)

Paper	Proposed Strategy	Complexity	Designed for Multi-hop Networks?	Approach used	Application
[3]	A-TXOP	Simple	Yes – Static nodes	Dynamically changing the TXOP limit values to reduce end-to-end delay.	Multimedia traffic
[3]	ReAP	Simple	Yes – Static nodes	Dynamically changing the priority of the messages to reduce end-to-end delay.	Multimedia traffic
[95]	DCC	Complex	Yes – Mobile nodes	Priority is dynamically adjusted based on the estimated per-hop delay to reduce end-to end delay. Modifies the routing table.	Multimedia traffic
[94]	APHD	Complex	Yes – Static nodes	Cross-layer technique which modifies the packet header to store application end-to-end information requirements and changes the priority adaptively on a per hop basis to achieve this requirement.	Multimedia traffic
[96]	Dynamic Priority	Simple	Yes – Static nodes	The priority of the packet keeps changing based on the network conditions considering the application required QoS.	Multimedia traffic
[98]	APS	Simple	Yes – Static nodes	Adaptively adjusting the packet length.	Multimedia traffic
[9]	SBB and SAB	Simple	Yes – Static nodes	In one scenario frames are first selected for transmission and then the channel access functionalities are performed. In the other first back-off takes place then frames are scheduled. A round robin with DCF strategy and weighted round robin (WRR) strategy followed by variable back-off parameters are investigated.	Multimedia traffic
[91]	WF-EDCA	Simple	No – Single hop	Designed to provide proportional fairness by providing weighted,fair service among different ACs.	Multimedia traffic
[92]	DS-EDCA	Complex	No – Single hop	Uses a virtual clock that requires synchronization to provide strict priority and weighted fair service for the different ACs.	Multimedia traffic
[93]	FQ-EDCA	Complex	Yes – Static nodes	Designed to provide fair queuing. It classifies packets from the upper layer and then further classifies packets within each AC deepening on whether the packets are control packets or data.	Multimedia traffic
[59]	SFS	Simple	No – Single hop	The technique determines the number of packets in each queue and then calculates the required time to transmit these packets. This information is then used to schedule these packets.	Multimedia traffic
[103]	EDDRRBI and EDERR-BI	Complex	No – Single hop	The schemes dynamically adjust the back-off interval according to the packet priority, collision rate and the deficit count or allowance.	Multimedia traffic

Chapter 3

Schedule-before-contention Wireless Mesh Network Scheduling Strategies

3.1 Introduction

This chapter introduces the five priority scheduling strategies that have been developed to improve the QoS in SRSC multi-hop networks. The scheduling strategies first choose a packet for transmission from one of the priority queues and then perform the contention period to gain access to the channel and transmit the data. The different priority data queues in a node do not contend for the medium at the same time as is the case in EDCA and do not have an internal contention mechanism. Therefore, we call these strategies as schedule-before-contention (SBC) packet scheduling strategies.

Before the scheduling mechanism can be applied for all these SBC strategies, packets need to be classified and placed in the respective class queues. For all the proposed strategies that have different scheduling mechanisms, when data arrives at the MAC layer, the data is placed into one of the three priority data class queues depending on the application from which the data originates. The classes used in the proposed schemes are high priority data, medium priority data and low priority data. Different applications use different transport layer protocols such as the transmission control protocol (TCP) or the user datagram protocol (UDP) that have different port numbers. Port numbers in the header of the frame are used to classify these packets. The algorithm for classification used is given in figure 3.1. After that, one priority queue is selected for transmission on the medium following a scheduling selection process. A packet from the head of line (HOL) is selected. After this, the medium is monitored for the AIFS and back-off period to determine if it is still idle and then the packet is transmitted on the channel. An overview of the SBC strategies in this work is shown in figure 3.2.

With the use of EDCA in multi-hop networks, there are high packet losses and collisions at heavy loads. The internal contention mechanism also reduces the chances of the lower priority data to gain access to the channel in the presence of higher priority packets in the other queues in a node. Five MAC layer scheduling mechanisms for the SBC strategies have been proposed. The scheduling mechanisms are called the adaptive weighted round robin (AWRR), the roulette wheel sampling (RWS), the RWS-AGE, the congestion control and fairness scheduling (CCFS) and the queue load control priority (QLCP) scheduling mechanism. The scheduling packet selection mechanisms are explained for each strategy in this chapter. All the strategies have been developed as enhancements to CSMA/CA and implemented at the MAC Layer. AWRR, RWS,

Algorithm 1 Packet Classification

```
PacketPrioirty = 1
for each incoming packet do
    check packet priority in the packet header
    if PacketPrioirty = 1 then
        Place packet in the High Priority Data Queue
    else
        if PacketPrioirty = 2 then
            Place packet in the Medium Priority Data Queue
        else
            if PacketPrioirty = 3 then
                Place packet in the Low Priority Data Queue
            end if

            Return PacketPrioirty
```

Figure 3.1: Packet Classification Algorithm.

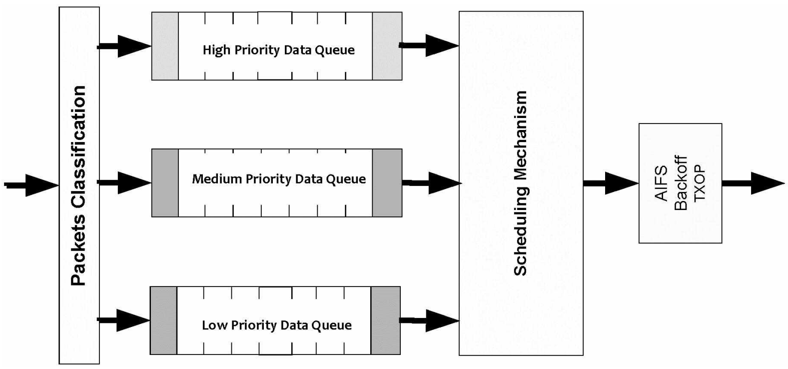


Figure 3.2: An overview of the schedule-before-contention packet scheduling strategies.

Table 3.1: Back-off contention window ranges

Priority Class	**Minimum CW value**	**Maximum CW value**
Low Priority	31	1023
Medium Priority	15	31
High Priority	7	15

RWS-AGE and CCFS have been proposed to reduce the overall collisions and hence reduce the overall packet loss in a network. QLCP has been proposed to reduce packet loss by trying to prevent packets from being dropped or starved by transmitting packets from queues with packets exceeding a threshold in bottle-neck nodes.

The back-off values for the CW used for the different priority packets is all the SBC strategies assigned are given in table 3.1.

3.2 Adaptive Weighted Wound Robin (AWRR) Scheduling Strategy

The AWRR scheduling mechanism aims at reducing packet loss as well as preventing starvation by increasing the number of lower priority packets that gain access to the channel compared to EDCA under heavy loads. This is expected to reduce packet loss as the lower priority data have larger CW sizes and therefore, have a lower collision probability than the higher priority data. The use of a weighted round robin (WRR) is proposed in this mechanism. A wheel is used which cycles from queue to queue depending on the number of slots allocated. This will ensure that after a certain number of high and medium priority data packets have been transmitted, the lower priority data is given access to the medium. The WRR is a very common CPU scheduling technique and has been investigated in many scheduling strategies implemented in different wireless standards such as in WiMax in [104], and in single-hop WLANs using IEEE 802.11 in [9], [91] and [92], but not for multi-hop WMNs. However, the performance of the WRR strategy in SRSC WMNs, using the IEEE 802.11 standard, has not been investigated before. With the proposed AWRR strategy, weights are assigned to the different priority queues with the higher priority data queues being assigned more transmission slots, depending on the assigned weights, than the lower priority data queues. The numbers of slots allocated to the different priority queues are application dependent, based on how much priority the different classes require. In this design, high priority (HP), medium priority (MP) and low priority (LP) queues are assigned slots from a wheel in the ratio of 5:3:2, respectively. The AWRR scheduling strategy is shown in figure 3.3. The queues that are empty are skipped. Under heavy loads, when more than one queue has data, the order in which the packets from the different queues are transmitted is fixed with high priority data transmitting first, then medium priority data and lastly low priority data and then the cycle restarts.

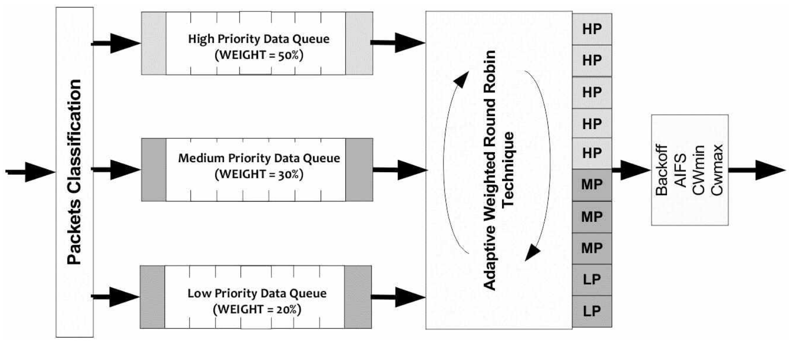


Figure 3.3: An overview of the AWRR scheduling strategy.

3.3 Roulette-Wheel Sampling (RWS) Scheduling Strategy

The same principle of AWRR to reduce packet loss is applied to this mechanism. This RWS scheduling mechanism also aims at reducing packet loss as well as preventing starvation by increasing the number of lower priority packets that gain access to the channel compared to EDCA under heavy loads by aiming to have different priority packets trying to access the channel by different multi-hop nodes at a time instance. This is expected to reduce packet loss due to the different CW sizes. The roulette wheel sampling selection scheduling mechanism works on the principle that the queue with a larger weight has a higher chance of getting selected and of transmitting its data compared to the queue with a smaller weight. In this strategy a queue for data transmission is probabilistically selected, based on the different weight values assigned. Let us consider J priority classes, each characterized by their weights $w_j > 0$ $(j = 1, 2, \cdots, J)$. The selection probability of the j_{th} priority class is given in equation 3.3.1.

$$p_j = \frac{w_j}{\sum_{j=1}^{J} w_j} \tag{3.3.1}$$

$$\sum_{j=1}^{J} p_j = 1$$

For illustration of the working principle of the roulette wheel sampling, let us consider a roulette wheel as shown in figure 3.4 with the different categories of priority assigned a size proportional to w_j $(j = 1, 2, \cdots, J)$. In this example, we assume high priority data has a selection probability of $p_1 = 0.5$, medium priority data has a selection probability of $p_2 = 0.3$ and low priority data has a selection probability of $p_3 = 0.2$.

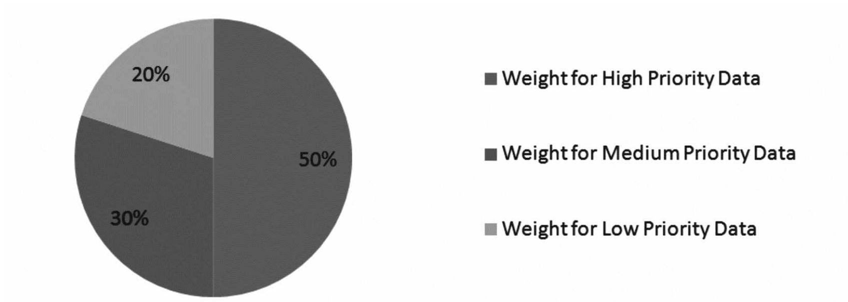


Figure 3.4: Weights assigned to the priority data classes using RWS.

The priority classes are then mapped to a continuous segment of a line, such that each priority class segment is equal in size to its selection probability. A number is then randomly chosen (r) as shown in figure 3.5. The AC sector that r points to is chosen for the transmission of the data provided that its transmission queue is not empty.

The flow chart for the operation of the proposed RWS-scheduling strategy is shown in figure 3.6. The stages of operation of the RWS scheduling are as follows:

1. Assign probability selection weight to each data priority queue.

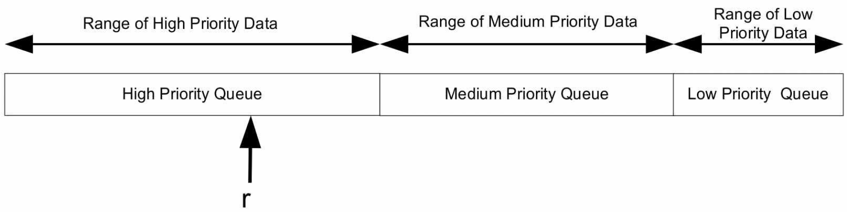


Figure 3.5: Roulette wheel sample selection.

2. The strategy then determines the size of the individual queues. If all of the queues have data, the original assigned weights in stage 1 are used. If all the queues do not have data, then the weight of the queues with data are added, the weights of the queues with data are normalised and assigned new weights. The queues with no data are assigned a weight of zero. This allows the strategy to be adaptive and adjust according to which queues have data.

3. The range values for each of the priority data classes are assigned over a scale.

4. A random number is generated between 0 and the maximum scale value. A packet is chosen for transmission from a queue from which the number generated falls in its range.

With this RWS mechanism packets are transmitted, based on probabilistic selection, giving a higher chance to packets having different CW ranges contending for the medium at a specific instant of time compared with AWRR. With AWRR a fixed number of packets are transmitted from each queue in a certain order under heavy loads, when more than one queue has data. This is expected to reduce the packet loss in RWS compared to AWRR due to the access of packets with different CW sizes for the back-off contending for the medium at a time.

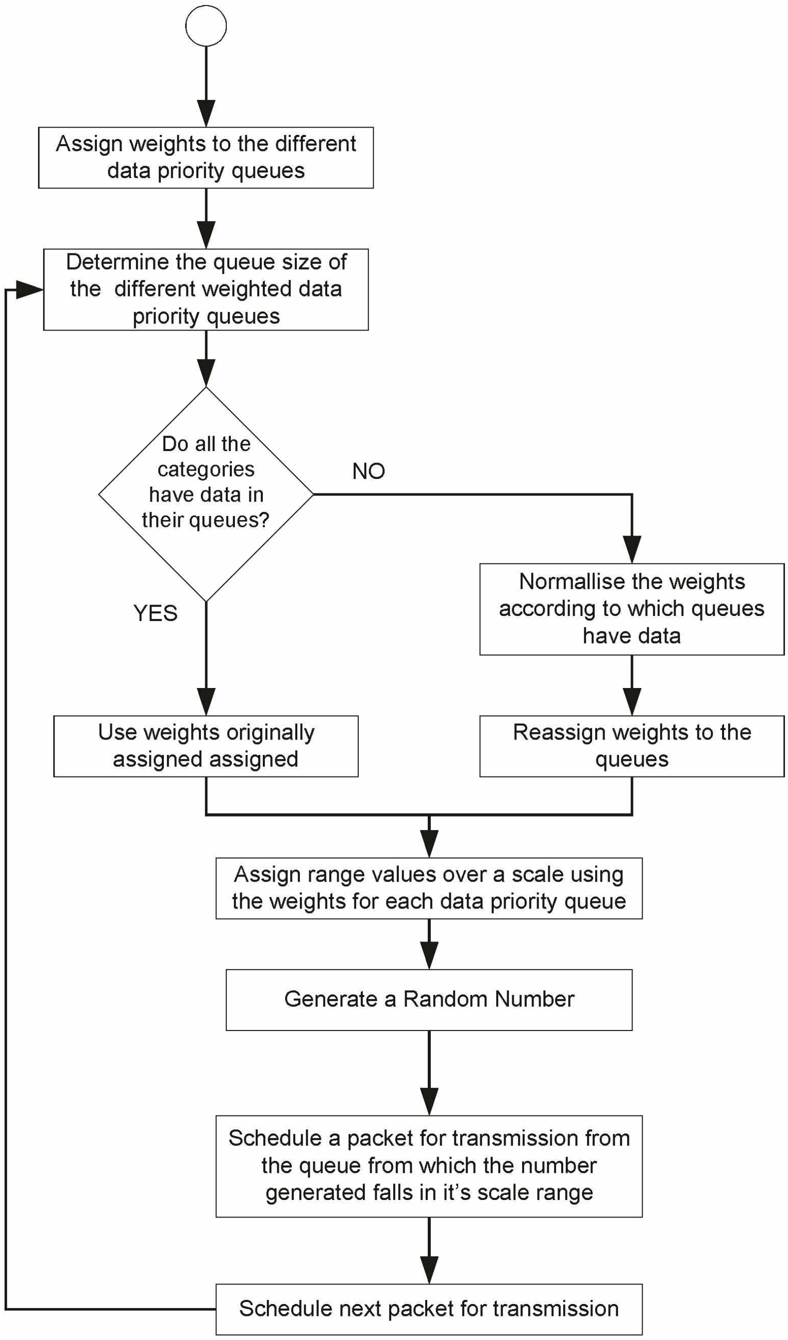


Figure 3.6: Flow chart for the proposed RWS mechanism.

3.4 RWS-AGE Scheduling Strategy

This RWS-AGE mechanism incorporates an age counter to the RWS mechanism. With this RWS-AGE mechanism, an age counter is used for each class data to determine how many consecutive packets of the same class are transmitted. The starvation counter prevents more than a predetermined number of packets from the same priority queue to be consecutively transmitted. In this work more than five packets from the same priority queue cannot be consecutively transmitted. The starvation counter value of five maintains the weight transmission probabilities assigned to the different queues and also ensures that, in a worst case scenario, starvation does not occur. With the AWRR strategy, the age counter is a default in the mechanism. A value of five is chosen as with AWRR, only five high priority packets can be transmitted consecutively as the wheel cycles. This will allow a direct comparison. A fixed number of the same priority packets cannot be transmitted if any other queue has data. The flow chart for the operation of the proposed RWS-AGE mechanism is shown in figure 3.7.

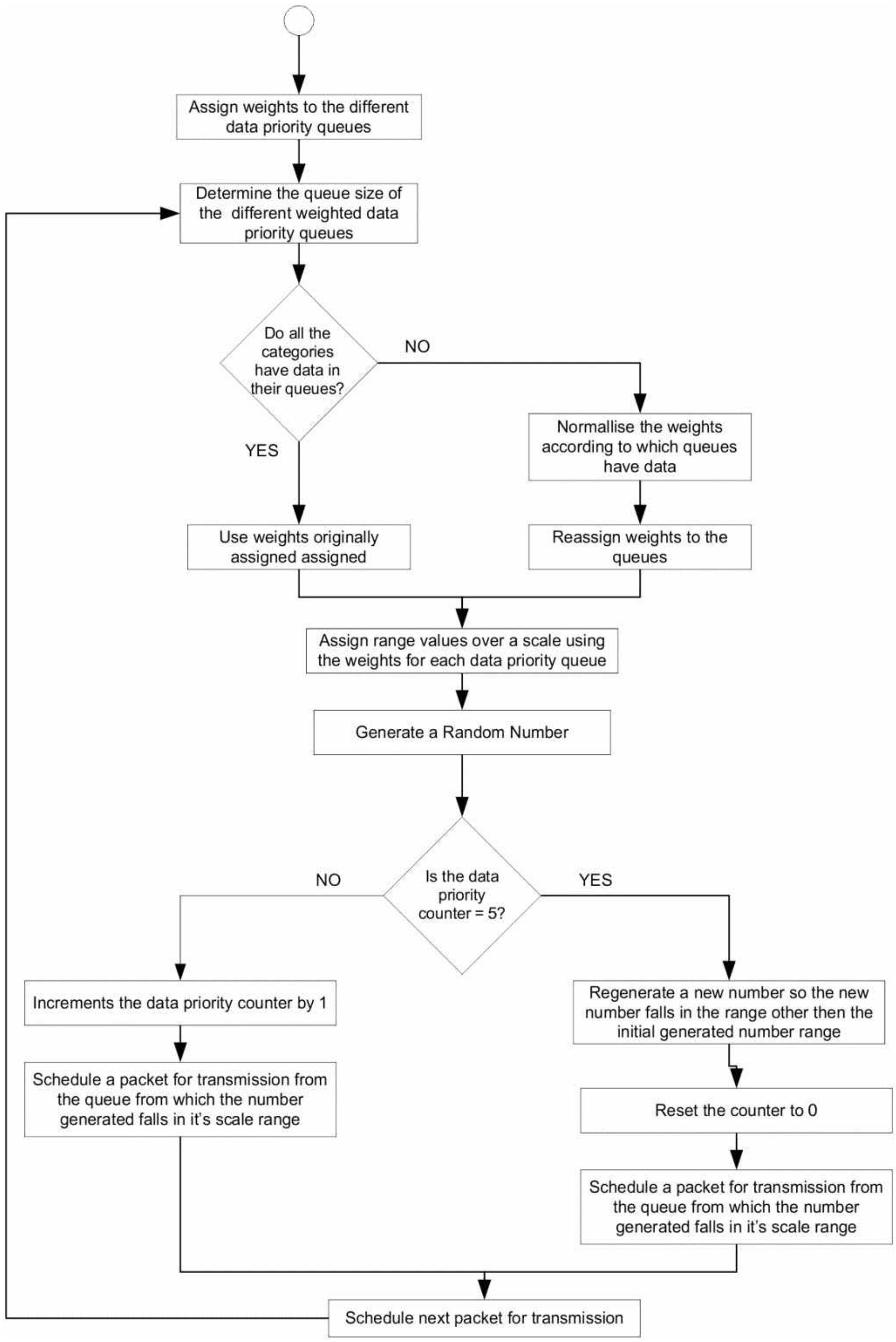


Figure 3.7: Flow chart for the proposed RWS-AGE mechanism.

3.5 Congestion Control and Fairness Scheduling (CCFS) strategy

If data in a priority queue start to increase and the packets are not given access to the channel quicker, there is a possibility of starvation. To address such cases, the congestion control and fairness scheduling (CCFS) mechanism is developed and investigated with a queue length measure added to the AWRR mechanism. With CCFS, the number of slots assigned to each priority queue is adaptive depending on the load level in each queue. The number of slots assigned to each priority queue is reduced from the ratio HP:MP:LP of 5:3:2 using 10 slots to 3:2:1 using 6 slots with fewer transmission slots assigned. This is done to allow other queues to gain access to the medium faster. The CCFS operates as follows:

1. The technique firstly determines which priority queues have data by checking the queue lengths.

2. If only one queue has data, data is scheduled from that queue. If more than one queues have data, either one of four flow charts as shown in figure 3.8 is followed, depending on which queues have data. In the case that only the medium priority (MP) queue and the low priority (LP) queue have data, than flowchart 3.8a is used. If the high priority (HP) and the low priority queues have data, then flowchart 3.8b is used. If the high priority and the medium priority queues have data, then flowchart 3.8c is used. If all the priority queues have data, then flowchart 3.8d is used.

3. The load threshold value for all the queues was set to 2, as having 3 packets waiting in the queue for transmission gives an indication that the queue has started to build up.

4. If only the medium priority and low priority queues have data, the queue length of the low priority queue is determined. If the load level is greater than the threshold, then the MP transmission slots assigned is set to 2, else it is set to 3. For every consecutive MP data transmitted, the MP age counter value is incremented by 1. If the MP age max value is reached, then a LP packet is immediately scheduled for transmission and the counter value is reset to 0. The remaining flow charts are interpreted in a similar way. The HP and MP age max counter values differ, depending on which queues have data. The counter values are chosen such that higher priority data can be sent out in preference to lower priority data.

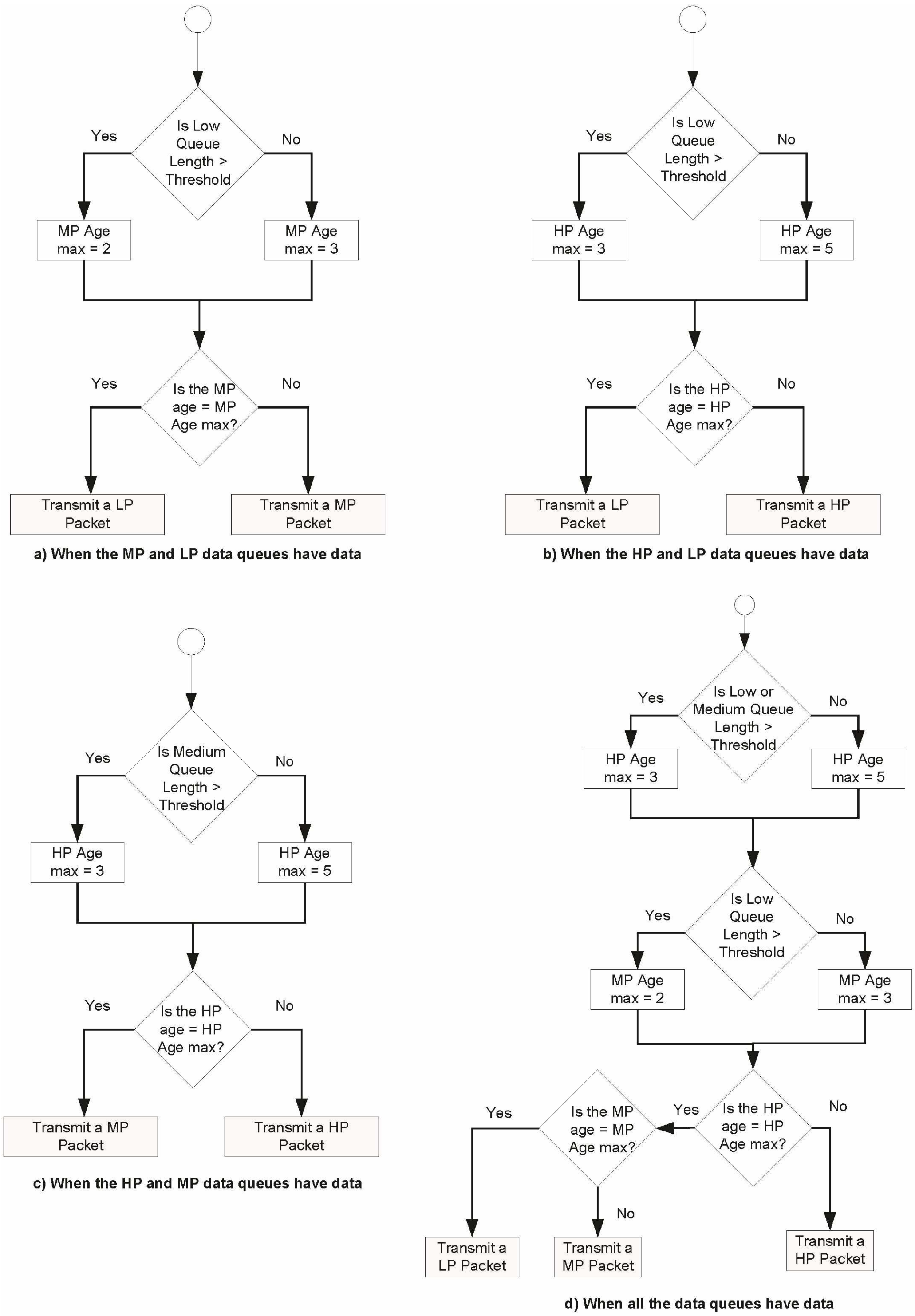


Figure 3.8: An overview of the CCFS scheduling mechanism.

3.6 Queue Load Control Priority (QLCP) Scheduling Strategy

QLCP has been proposed to reduce packet loss by trying to prevent packets from being dropped from queues that are getting too long as well as preventing starvation of data in queues that are becoming long. In many network layouts, such as those where the backhaul WMN architecture is used, the edge or gateway nodes are subjected to more traffic load and congestion as they constitute the entry point of that part of the network domain to the backbone network. In this layout queue size is used as a metric for detecting congestion and for load control. This strategy relies on the principle that the queues that have more data should be allowed to transmit their data first, in accordance with the assigned priority. This means that queues which have more data than the threshold value and higher priority data are given preference to queues with lower priority. The QLCP operates in a tree structure as shown in figure 3.9. The load threshold value is adaptive and keeps changing depending on the total load in all the queues and is calculated at 30% of the combined load from all the queues to prevent queues from getting congested. A value of 30% is used as when all the queues have one packet, the HP queue must transmit. The calculated threshold value will is 0.9, which is less than the queue length value.

The QLCP strategy determines the length (load) of the different priority queues, starting with the highest priority queue. It checks the load to determine if it is greater than the threshold value, as illustrated in levels 1 to 3 in figure 3.9. If it is higher than the threshold value, it is scheduled for transmission. This assists congestion control and allows transmitting data from long queues instead of dropping them when the queue is full. In EDCA queue length does not influence the queue for transmission and under heavy loads chances are therefore, higher for lower priority data to get starved, whereas in QLCP the queue length plays a significant role in queue selection for transmission. In QLCP, as illustrated for levels 4 and 5 in figure 3.9, a packet from a higher priority queue with data is scheduled for transmission provided all the queue lengths are less than the threshold value. The algorithm for QLCP is presented in figure 3.10.

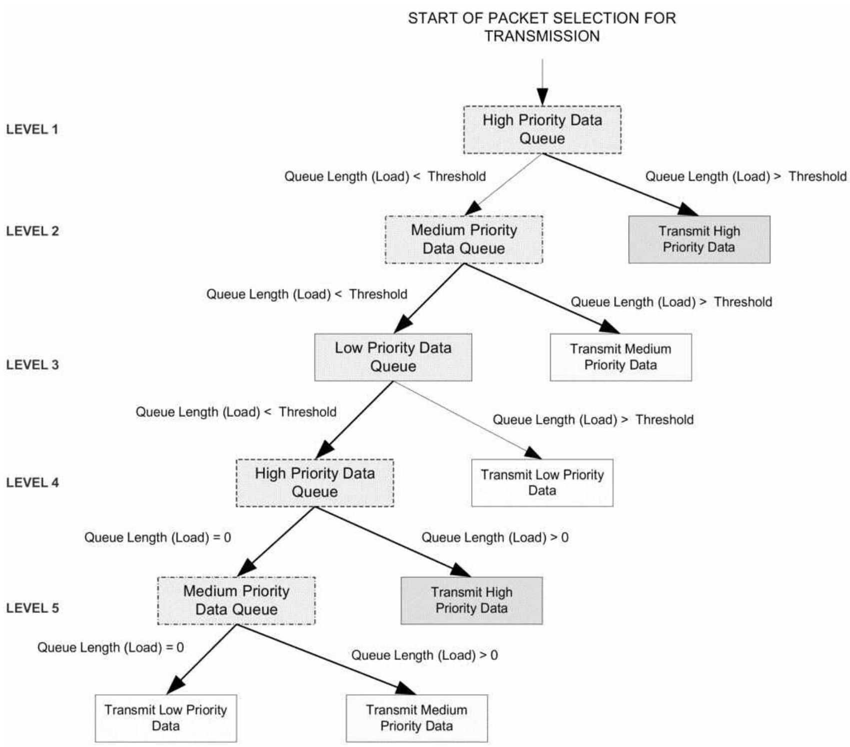


Figure 3.9: QLCP packet scheduling strategy

Algorithm 1 QLCP Scheduling Scheduler

$HPQueueSize = 0, MPQueueSize = 0, LPQueueSize = 0$
for each data priority queue **do**
    determine the queue size
    if $HPQueueSize > Threshold$ **then**
        Transmit a High Priority Data Packet
    else
        if $MPQueueSize > Threshold$ **then**
            Transmit a Medium Priority Data Packet
        else
            if $LPQueueSize > Threshold$ **then**
                Transmit a Low Priority Data Packet
            else
                if $HPQueueSize < Threshold\ AND\ HPQueueSize > 0$ **then**
                    Transmit a High Priority Data Packet
                else
                    if $MPQueueSize < Threshold\ AND\ MPQueueSize > 0$ **then**
                        Transmit a Medium Priority Data Packet
                    else
                        if $LPQueueSize < Threshold\ AND\ LPQueueSize > 0$ **then**
                            Transmit a Low Priority Data Packet
                        end if

                        Return *Packet Prioirty Queue for Transmission*

Figure 3.10: QLCP scheduling strategy algorithm.

3.7 Comparison between the Proposed Schedule-before-Contention Scheduling Mechanisms

The scheduling strategies proposed in this research deviates from EDCA as only one packet queue is chosen for each transmission, from all the priority queues for contention for the medium. Also, with DCF, schedule-before-contention is carried out as it only has one queue. With the proposed strategies, no internal collisions take place as only one packet from each of the queues is scheduled at a time. Internal collisions take place with EDCA, as can be seen in figure 2.5, as each queue behaves as a virtually separate node and this contention is performed separately in parallel. The proposed strategies discard the internal collisions mechanism as the data from the different priorities queue are not contending for the medium at the same time. The advantage of removing the internal contention mechanism is that it prevents internal collisions from taking place and also prevents starving lower priority data. The AWRR, RWS-AGE and CCFS strategies also maintain an age counter to detect the transmission of consecutive data from the priority class. This is a further measure in place to avoid starvation. Figure 3.11 presents the complete basic overview model of all the proposed SBC scheduling strategies based on different scheduling mechanisms.

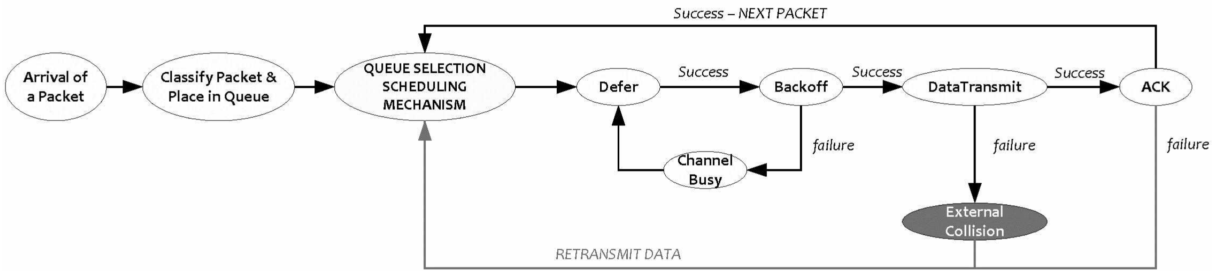


Figure 3.11: Medium access flowchart for the proposed scheduling-before-contention strategies.

Table 3.2 presents a comparison of the proposed scheduling strategies and the baseline contention based strategies namely EDCA and DCF. All the proposed strategies perform classification and sort data into the different queues and consider which queues have data. Both QLCP and CCFS base their operation on the load level in each queue, whereas AWRR, RWS-AGE and CCFS consider the number of packets sent out consecutively from the same queue while another queue also has data.

3.8 Conclusion

This chapter has presented an overview of the operation of the proposed SBC strategies using different mechanisms and has also provided a brief comparison of the strategies with EDCA and DCF. Whereas DCF operates in a FIFO fashion, AWRR uses a round robin technique. RWS and RWS-AGE employ a random selection technique based on weights assigned to the respective queues. The CCFS operates in a flow chart fashion, based on load and age of the packets in the queues. The operation of QLCP takes into account the load level in the respective queues based on priority. All the proposed techniques have been developed for wireless multi-hop networks.

Table 3.2: Comparison Between the Scheduling Mechanisms.

	EDCA	**DCF**	**AWRR**	**RWS**	**RWS-AGE**	**CCFS**	**QLCP**
Data Differentiation	YES	NO	YES	YES	YES	YES	YES
Consider the Load Level?	NO	NO	NO	NO	NO	YES	YES
Considers the AGE of packets?	NO	NO	YES	NO	YES	YES	NO
Adaptive to which queues have data?	YES	NO	YES	YES	YES	YES	YES
Parameters that can be adjusted to different application needs	NONE	NONE	Round Robin Slots assigned to each priority category	Weights assigned to each priority data queue	1. Weights assigned to each priority data queue, 2. Age	1. Queues Threshold Values, 2. Age	Queues Threshold Values
Scheduling	Contention Based	FIFO	Weighted Round Robin - Deterministic	Random	Random	Predefined flow chart based-Deterministic	Load level and Priority Based, Deterministic

Chapter 4

Homogeneous Configured Network Layout Schemes

4.1 Introduction

This chapter presents an overview of the homogeneous configured network layout schemes concept, the motivation for the experiments, the simulation environment, simulation parameters and the performance metrics used to analyze the performance and to do conduct a comparative analysis of the schedule-before-contention (SBC) scheduling strategies in the homogeneous configured network layouts, the results and the discussion of the results. Homogeneous configured network layout schemes are the layouts in which all the nodes in the network are assigned the same scheduling strategy.

4.2 Motivation For Experiments

Homogeneous configured network layout scheme experiments with the different SBC strategies each using different scheduling mechanisms are set up to investigate the following research questions, based on the hypotheses:

- Question 1: Does replacing the internal contention mechanism in EDCA with a weighted round robin scheduling mechanism reduce packet loss?
- Question 2: Which SBC mechanism results in the lowest packet loss and end-to-end delay?
- Question 3: In terms of packet loss, is a SBC strategy better than having an internal contention mechanism as in EDCA?
- Question 4: Does the use of TXOP in SBC strategies improve performance in terms of lowering packet loss and end-to-end delay?

4.3 Simulation Setup

In order to obtain a realistic picture of the behaviour of the scheduling protocols, it is imperative to test them on different topologies with different load levels. These strategies are mainly developed for SRSC resource constrained implementations and therefore, the simulation parameters were set to comply as far possible with such environments. There are many commonly used discrete event simulators (DES) being used to analyze the behaviour of protocols in WMNs. Some of these include OPNET, NS-2, NS-3, OMNeT++, QualNet and Glomosim. In this investigation, OMNeT++ was used with the INETMANET framework, due to its open source nature as it allows the use and modification of already available code. OMNeT++ is a very popular open source application discrete event simulator for simulating communication networks. OMNeT++ comes with many framework units and modules developed for computer networks [39]. The modules in OMNeT++ are programmed in C++ and then assembled into larger components and models using a high-level language (NED). The INETMANET library offers detailed models of radio propagation, implementation of various protocols of wireless network from the different OSI layers and applications, making it possible to simulate WMNs [106]. OMNeT++ with the INETMANET framework has been used in many EDCA based studies such as in [107].

4.4 Simulation Propagation Models

In this research, two propagation models were used, namely the free space model and the two-ray ground model. The free space model was used for the analytical model presented in chapter 7. The two-ray ground model was used in the simulations. A brief overview of these two models is presented, highlighting the reasons why each of these models was chosen.

Propagation characteristics of the environment are important in order to investigate the operation of the wireless scheduling strategies. The propagation models developed in wireless communication systems theory, focus on predicting the average received signal strength at different distances from the source and emulate the radio characteristics of a given environment [108].

4.4.1 Free Space Model

The free-space model which is also known as the Friis equation is a model where the received power depends on the transmitted power, the gain of the antenna and the distance from the transmitter. The received power weakens with an increase in distance [108,109]. The equation for the received power with the free space model is given in equation 4.4.1.

$$P_R = P_T G_T G_R (\frac{\lambda_{WL}}{4\pi d})^2 \tag{4.4.1}$$

where
P_R is the received power
P_T is the transmitted signal power
G_T is the antenna gain of the transmitter
G_R is the antenna gain of the receiver
λ_{WL} is the wavelength
d is the distance between the transmitter and receiver

The free-space model assumes ideal channel conditions with no losses taking place due to channel errors and hidden-terminal nodes. It assumes that there is only one line of sight (LOS) path between the sender and the receiver. We assume ideal channel conditions in the analytical model and therefore, in our simulations for comparing simulated results to the analytical model presented in chapter 7, we use this free-space model.

4.4.2 Two Ray Ground Model

The Two Ray Ground Model is an extension of the free space model to consider a case with a reflected signal or two paths for the signal from the transmitter to the receiver. This extension includes the ground reflection in the propagation. The free space model assumes that only one single path from the sender to the receiver is present. In some environmental settings, such as in rural areas, the signal can reach the receiver through multiple paths that take place due to reflections, refraction and scattering. This model assumes that the signal from the source reaches the receiver through two paths. One being the direct Line of Sight (LoS) path, and the other the reflected path [110,111]. This model assumes that the signal from the source reaches the receiver through two paths. One being the direct Line of Sight (LoS) path, and the other the reflected path [108,109]. The equation for the received power with the two ray model is given in equation 4.4.2.

$$P_R = P_T G_T G_R (\frac{h_t h_r}{d^2})^2 \tag{4.4.2}$$

where
P_R is the received power
P_T is the transmitted signal power
G_T is the antenna gain of the transmitter
G_R is the antenna gain of the receiver
λ_{WL} is the wavelength
d is the distance between the transmitter and receiver
h_t is the height of the transmitter
h_r is the height of the receiver

Assuming the application to resource constrained networks in rural areas in Southern Africa such as in Botswana and South Africa are mostly less dense with few buildings and mostly open wide areas making the two-ray model more suitable. In the simulations of the proposed scheduling strategies, we use this propagation model as in rural areas, these two rays exist predominantly, i.e. direct rays and the reflected rays. Moreover, the numbers of obstacles are fewer in rural areas compared to urban areas and as such, the two ray model was found to be more appropriate for rural areas.

4.5 Simulation Topologies

Grid and Line topologies are used in our simulations as a node can communicate with its neighbours, provided Omni-directional antennas are used and if the neighbors are in their transmission range. Grid topologies provide more contention links than line topologies, resulting in high contention which is helpful in assessing the performance of the strategies under high channel contentions. The studies in [112], [113] and [114] have proposed the suitability of grid topologies for the testing of WMN protocols. Lines topologies with high data rates as well as other nodes in communication simultaneously increase contention as well in the network and are also acceptable for testing. Random topologies are not used, as some nodes may then be placed outside of the coverage area of the node. Flow in the middle line topologies with other flows as well as flows that cross current flows are normally used in research projects to assess scheduling protocols in mesh networks [20]. Since the nodes in the topologies are not mobile and stay static, the grid and line topologies are suitable to access the performance of these scheduling strategies.

Table 4.1: Overview of the test case topologies.

	Description	Data flow links
TOPOLOGY 1	5 by 5 square grid	Data transmitted on different mesh links in multi-directions. Another flow crosses the main flow from the source to destination.
TOPOLOGY 2	5 by 5 square grid	Data transmitted with 2 other flows crossing the main flow from the source to destination.
TOPOLOGY 3	Line topology with 5 hops	The furthest hop node sends data to the sink node in one direction while two other nodes send data to another node in the network.

Three backbone WMN topologies were used to compare the performance of these scheduling strategies in homogeneous configured network layouts. Two of the topologies are square grid topologies and one is a line topology. The topology diagrams are shown in figure 4.1 and table 4.1 presents an overview of these test topologies. All the nodes forward incoming data as well as generate and transmit their own data. Every node in these topologies can communicate with its neighbours, falling within the coverage range of omni-directional antennas when used. Research results were obtained by measuring the data received at the destination node. The access points (AP) generate the data that flows in the network and receives the data sent for testing the scheduling strategies. In topology 1, domain 1 sends data to domain 3 and domain 2 sends data to domain 4. In topology 2, domain 1 sends data to domain 2 as well as 2 other randomly selected nodes are sending data to other randomly selected random nodes in a network. In topology 3, a line topology is used where the furthest node sends data to the other end. Two other nodes in the network also send data to another node in the network. All the topologies have one or more flows in the middle of the flow or that crosses the main flow from the source to the destination to assess the strategy in a mesh layout.

4.6 Other Simulation Parameters

Table 4.2 presents the simulation setup. All the nodes were configured with the IEEE 802.11g standard at the MAC and physical layer with nodes transmitting the MAC service data units (MSDU) at 54 Mbps and operating in the 2.4 GHz band. The OLSR routing protocol was used as in HWMP, a proactive protocol is used between static nodes. Also in the test scenarios and network layouts, no network changes are taking place. The same parameters for the priority queues as EDCA were used.

In many telemetry networks, packet sizes are usually between 60 bytes and 600 bytes instead of having one very big packet such as 1 kilobytes [115–117]. The reason for this is also that the information carried in telemetry packets is small compared to multimedia packets. Smaller packets have a lower probability of collision [118].

User Data Protocol (UDP) data packets at the transport layer were used in our simulations for the three types of priority data. UDP packets at the transport layer having a size of 512 bytes were used for the reason that UDP applications such as Trivial File Transfer Protocol (TFTP) and Domain Name Systems (DNS) use a default packet size of 512 bytes. As UDP does not establish connections between the source and destinations (connection-less) and as there is no re-transmission of lost packets, it was preferred above TCP [40]. This helps to detect the unreliability performance of the network at the lower layers caused by packet loss measures. The advantage of using UDP packets is that the re-transmissions that will result at the MAC layer as CSMA/CA re-transmits a packet due to lost packets or unacknowledged packets will be purely

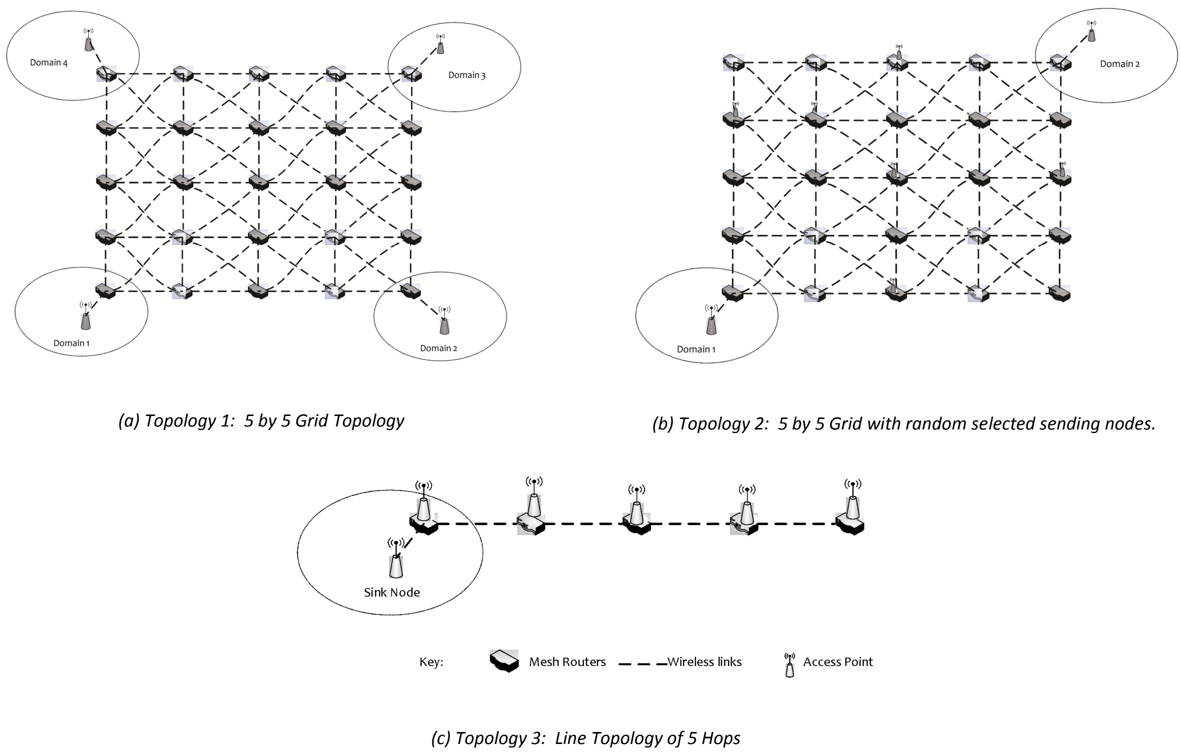


Figure 4.1: Topology test cases.

Table 4.2: Simulation Environment

Simulation Time	300sec
Nodes Separation	300m
Propagation Model	Two Ray Ground Model
Transmission Power for the Mesh Nodes	2mW
Routing Protocol	OLSR
Data Bit Rate	54Mbps
Basic Bit Rate	6Mbps
Transport Protocol	UDP Packets
Application Data Categories	High, Medium and Low Priority
Packet Size	512bytes
Confidence Interval	95%
Seed number separation	100000

due to the behavior or happenings at the MAC layer.

Different data transmission rate test cases were run on these topologies for the priority data sets given in table 4.3 by applying the different scheduling strategies. These different data transmission rate test cases cause different load levels of different data priority globally in the network and as such, thus serving as an assessment of the performance of the different strategies over differing load levels. The normalized offered load used for the testing was between 0.3 and 0.9. With a value of 0.5, there is always a packet in the collision domain either in the queue or being processed for transmission. With a load greater than 0.9, the system becomes unstable and the loss in performance after that is due to the network condition and not due to the scheduling strategy.

Table 4.3: Different load level test scenarios.

Load Level	Normalised Offered Load
Low	0.3
Medium	0.6
High	0.9

Each simulation was repeated five times with each simulation using a different seed number, generated by the random number generator utility in OMNeT++. OMNeT++ uses a deterministic algorithm, called the Mersenne Twister RNG, to generate random numbers and initialises it to the same seed. The results presented in the next section are averaged and plotted with error bars showing the 95% confidence interval.

In EDCA, TXOP has been applied after the contention parameters. The advantage of TXOP is that multiple packets from the same queue can be transmitted without the need of continuously performing the contention period. Experiments are carried in our homogeneous configured network layout schemes with and without the use of the TXOP.

4.7 Performance Metrics

To evaluate the performance of the proposed scheduling strategies presented in chapter 3, the end-end-delay, packet loss, number of collisions and Jain's Fairness Index metrics are used. The details of these performance metrics are presented below. The performance of the proposed scheduling strategies is compared with the EDCA and DCF baseline scheduling strategies.

1. **End-to-End Delay:** This is a very common metric used in determining the QoS. These metric measures the average time it takes a packet to reach the destination after being sent from the source. Delays can be caused due to re-transmissions between the hop links, route discovery, data queuing, propagation delays, SIFS, acknowledgement messages, AIFS and packet transmission time. The is the time it takes for a packet to travel from the source to the destination over the network.

2. **Percentage Packet Loss:** This is also a very common metric used in determining QoS. This metrics counts the number of packets sent at the transmitter and also counts the number of packets received at the destination. These values are used to calculate the packet loss as given in equation 4.7.1 [42]:

$$PacketLoss = \frac{(N_t - N_r) * 100}{N_t} \tag{4.7.1}$$

   where
   N_t is the number of packets transmitted
   N_r is the number of packets received

3. **Number of Collisions:** With CSMA/CA, a node contends for the medium. A collision can occur if two nodes transmit at the same time with data of any priority class. The number of collisions experienced by the different scheduling schemes in the different test cases

and different topologies is measured and then used to calculate the number of collisions per *ms*. This value indicates the bandwidth usage on the network as with more collisions, the bandwidth is wasted as the packets have to be re-transmitted.

4. **Jain's Fairness Index (JFI):** There are many metrics proposed in literature to measure fairness. These among others include Min-Max, Entropy, and Jain's Fairness Index. The most common fairness measure metric found in literature for measuring fairness in communication networks is the JFI which is regarded as the de facto standard in communications. It measures how fair or unfair the resources are shared among the competing hosts by giving a value between 0 and 1. Since the strategies aim at preventing starvation, which will directly improve fairness, the JFI metric will be a good measure to determine if the fairness is improved or not. With JFI, a value between 0 and 1 is always obtained. A value close to 1 indicates the highest fairness while those close to 0 indicate the most unfair [43]. Equation 4.7.2 calculates the fairness.

$$f(y_1, y_2, \cdots, y_J) = \frac{(\sum_{j=1}^{J} y_j)^2}{J \sum_{j=1}^{J} y_j^2} \tag{4.7.2}$$

where

y_j is the normalised throughput of the the j^{th} priority class
J is the number of priority queues in a node

$0 \leq \mathrm{f}(y_1, y_2, \cdots, y_J) \leq 1$

4.8 Results and Discussion

4.8.1 Performance of the strategies without the use of TXOP

This section presents a performance analysis of DCF, EDCA, AWRR, RWS, RWS-AGE, CCFS and QLCP scheduling strategies. All the proposed strategies namely AWRR, RWS, RWS-AGE, CCFS and QLCP are SBC strategies with novel scheduling mechanisms. A comparison in performance in made between these MAC layer scheduling strategies and mechanisms in terms of packet loss, number of collisions, end-to-end delay and Jain's Fairness Index.

4.8.1.1 Collisions

Figure 4.2 presents the graphs showing the total number of collisions in the network per *ms* with the different scheduling strategies in topology 1. Figure 4.3 presents the graphs showing the total number of collisions in the network in topology 2 and figure 4.4 presents the graphs showing the total number of collisions in the network in topology 3. In the charts, LL refers to low load, ML refers to medium load and HL refers to high load.

For all the test topologies, EDCA experienced the most number of collisions under low, medium and high loads in all topologies except that CCFS experienced the most collisions under high load in topology 1. DCF experienced the least number of collisions under all loads in all the tested topologies. The AWW, RWS and RWS-AGE mechanisms experienced fewer collisions than EDCA, CCFS and QLCP, while were more than DCF on average. It can be seen that having larger CW value ranges for the back-off process reduces the collision probability. When a number is selected over a larger range by two nodes, the chances are lower of them selecting the same back-off number as compared to when the range is smaller. For all the priority classes, DCF uses a larger CW range for the back-off ($CW_{min} = 31$ and $CW_{max} = 1023$) compared to EDCA, RWS, AWRR, RWS, RWS-AGE, CCFS and QLCP which use the values presented in table 3.1. Increasing the number of lower priority data packets with larger CW ranges or other packets with larger CW ranges results in a decrease in packet loss. Lower collisions overall helps to utilize the channel bandwidth more efficiently. Therefore, the AWRR, RWS and RWS-AGE scheduling mechanisms on average utilise the bandwidth more efficiently for heterogeneous data priority scheduling. On average the RWS-AGE SBC mechanism experienced the lowest collisions compared to AWRR and RWS.

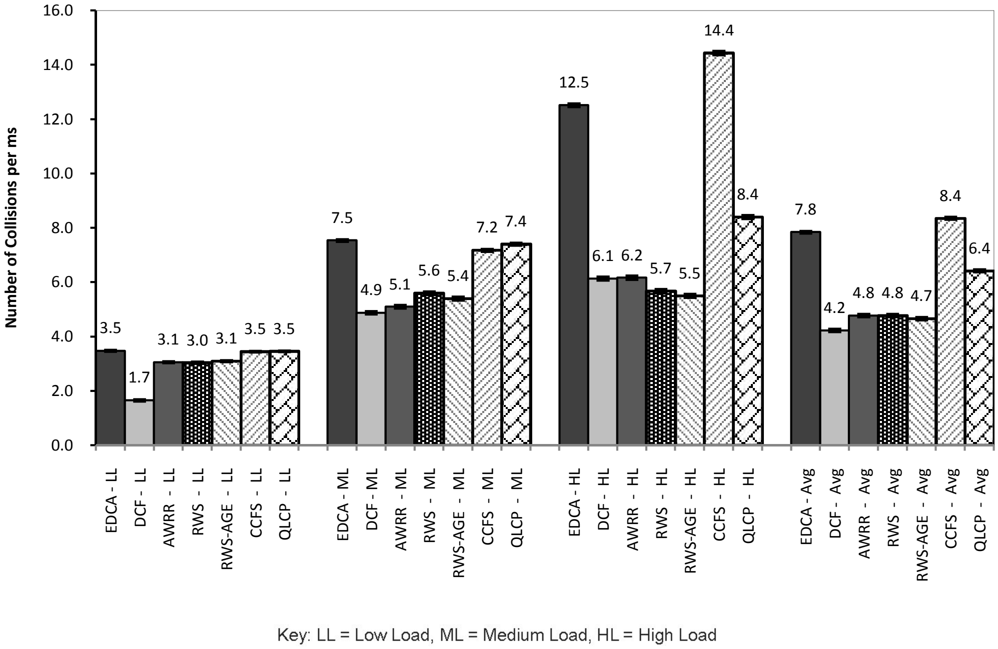


Figure 4.2: Average Number of Collisions with the different scheduling strategies in Topology 1.

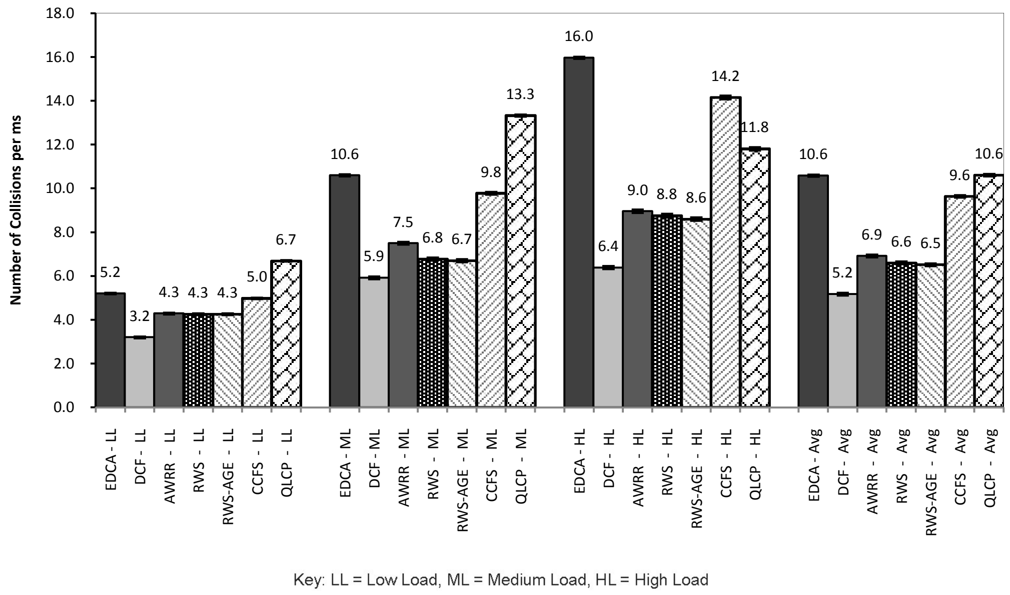


Figure 4.3: Average Number of Collisions with the different scheduling strategies in Topology 2.

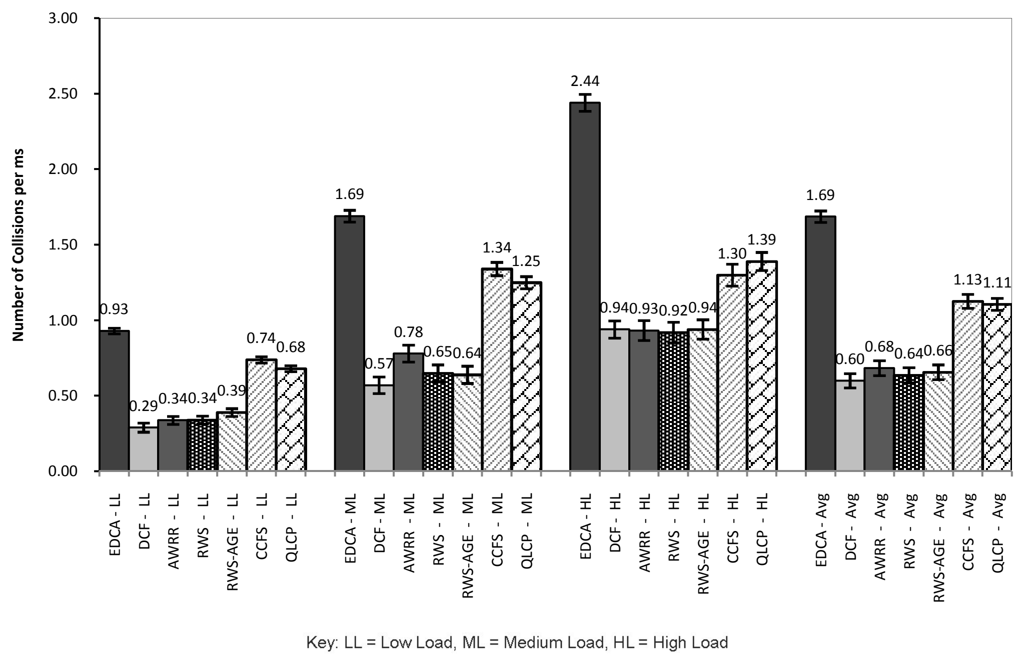


Figure 4.4: Average Number of Collisions with the different scheduling strategies in Topology 3.

4.8.1.2 Packet Loss

Table 4.4 presents a summary of the packet loss for all the different priority data classes over the different network loads in topology 1. Table 4.5 presents a summary of the packet loss in topology 2 and table 4.6 presents a summary of the packet loss in topology 3. Figure 4.5 presents packet loss with the different priority queues in topology 1 and figure 4.6 presents the average calculated packet loss experienced in topology 1. Figure 4.7 presents packet loss with the different priority queues in topology 2 and figure 4.8 presents the average calculated packet loss experienced in topology 2. Figure 4.9 presents packet loss with the different priority queues in topology 3 and figure 4.10 presents the average calculated packet loss experienced in topology 3.

DCF experienced less packet loss than EDCA for low and medium load scenarios in all the test topologies. In high load scenarios, the packet loss with DCF is higher than EDCA. DCF use larger CW range values which reduce the collision probability. EDCA on average experiences high packet loss under all load scenarios.

With the proposed scheduling strategies, one packet at a time is scheduled for transmission depending on the selection mechanism. With EDCA, if two or more queues have data to transmit, the AIFS and back-off periods are performed concurrently. If two data packets from different queues finish this period at the same time, it results in an internal collision starving the lower priority data. With the proposed schemes, there are no internal collisions and therefore, does not disadvantage the lower priority data, but is dependent on the scheduling mechanism. It can be seen that with all topologies, the AWRR, RWS and RWS-Age mechanisms experienced the least packet loss with slight variations between them. Their packet loss was lower than DCF, EDCA, CCFS and QLCP under medium and heavy loads. For low loads, DCF experienced the least packet loss.

This observation answers the research question on whether replacing the internal contention mechanism in EDCA with a weighted round robin scheduling mechanism would reduce packet loss. The results clearly show less packet loss than EDCA in all the test topologies. There is a packet loss reduction of 9.6% on average with AWRR over EDCA in topology 1, 11% on average with topology 2 and 18.4% on average in topology 3 under high loads. There is a packet loss reduction of 9.6% on average with RWS over EDCA in topology 1, 10.8% with topology 2 and 24.5% in topology 3 under high loads. There is a packet loss reduction of 10.8% with RWS-AGE on average over EDCA in topology 1, 14.8% with topology 2 and 21.1% in topology 3 under heavy loads. The weighted round robin mechanism cycle from the high priority queue to the lower priority queues and transmits more packets from the higher priority queue if it has data. The lower priority data is guaranteed to get access to the cycle when the wheel reaches the lower priority transmission slots.

Higher packet losses are observed in topology 2 compared to topology 1. This is due to the higher contention level in topology 2 with more nodes trying to gain access to the medium compared to topology 1. The CCFS and QLCP mechanisms experienced higher packet loss than the AWRR, RWS and RWS-AGE mechanisms under heavy load scenarios. The CCFS strategy changes the number of slots assigned to the different queues when the load exceeds the threshold value in any queue in the mechanism. This ends up lowering the overall transmission probability of the lower priority data and therefore, resulting in starvation of the lower priority data. Although it was expected that reducing the transmission wheel size by assigning fewer slots when the load increases will help with transmitting packets quicker from the queue that is becoming longer, the performance shows it actually leads to more starvation of lower priority data. This was not anticipated. The performance with CCFS in all saturated conditions is shown to favour transmissions of high priority data. If for example, there are HP and LP data packets in a node and the queue length of LP data is more than the threshold, this will imply that after this detection, a further three HP packets can be transmitted before the LP data is given a chance to transmit. On average, CCFS experiences less packet loss than EDCA. The QLCP mechanism experiences less packet loss than CCFS in high load scenarios. QLCP is designed to transmit packets from the longer queues first in the order of their priority. Therefore, with QLCP, the lower priority data have a higher chance of accessing the medium under heavy loads. Overall in the network, more lower priority packets will be in transmission than EDCA which lowers the collision probability on the network. The lowering of the collision probability arises as a result that lower priority data use wider CW ranges as stated earlier.

These results provide support to our research question on whether a SBC strategy is better than having an internal contention mechanism as in EDCA in terms of lowering packet loss. In all the SBC strategies, over all the test topologies and load levels, lower packet loss on average is observed with the SBC mechanisms than using EDCA. Some mechanisms do starve lower priority data under heavy load, but the loss experienced on average is not more than that which is experienced with EDCA. CCFS experiences 12.2% less packet loss on average than EDCA in topology 3, 9.1% in topology 2 and 6.3% in topology 1 under high loads. QLCP experiences 16.1% less packet loss on average than EDCA in topology 3, 10.3% in topology 2 and 6.4% in topology 1 under high loads.

The results in this section have shown that the design of the scheduling strategy can have a significant impact on the QoS achievable in terms of packet loss in WMNs. This supports our hypothesis as well that the scheduling algorithm has a global affect on the achievable QoS. CCFS tends to starve lower priority data as well even though there is no internal contention mechanism. In CCFS, a threshold value for queue length is used such that the age counter is made smaller if the lower priority queue is higher than this threshold value. Higher priority packets to the number of the age counter can still be transmitted after the queue length of the lower priority is detected to be higher than the threshold. The AWRR, RWS and RWS-AGE scheduling mechanisms are adaptive and change the number of slots or weights for each priority class depending on which queues have data. The channel access probabilities are only proportional to which queues have data and not queue load. On average over all the test topologies, the RWS-AGE mechanism experienced lower packet loss than the RWS mechanism in topologies 1 and 2 under high loads. The RWS-AGE mechanism has an addition age counter. This ensures under all conditions, if any other queue has data, if five packets from any one queue are transmitted, another queue is given the opportunity to transmit. Increase in the transmission of lower priority data works in the favor of reducing packet loss due to lower priority data having larger CW sizes. Overall, the RWS-AGE mechanism experiences less packet loss than AWRR under high loads for all test topologies. With AWRR only the number of packets equivalent to the number of slots assigned to each priority category can be transmitted consecutively if another priority queue has data. This therefore also behaves as having a default counter. To determine if the RWS-AGE performs statically better than AWRR, the paired T-Test in the heavy load cases is carried making a null hypothesis that the means are the same and an alternate hypothesis that the means are different. The calculated T value is 11.03 which is greater than the T critical value of 2.77 for HP data in topology 2. We therefore, reject the null hypothesis and show statically that RWS-AGE experiences lower packet loss than AWRR on average. This statistical test provides the answer to our research question on which SBC mechanism results in the lowest packet loss. The application of RWS-AGE on average results in the least packet loss with a 95% confidence level. The random probability weight assigned selection mechanism with an age counter performs better than a weighted round robin wheel for transmission of heterogeneous data.

Table 4.4: Packet Loss in Topology 1.

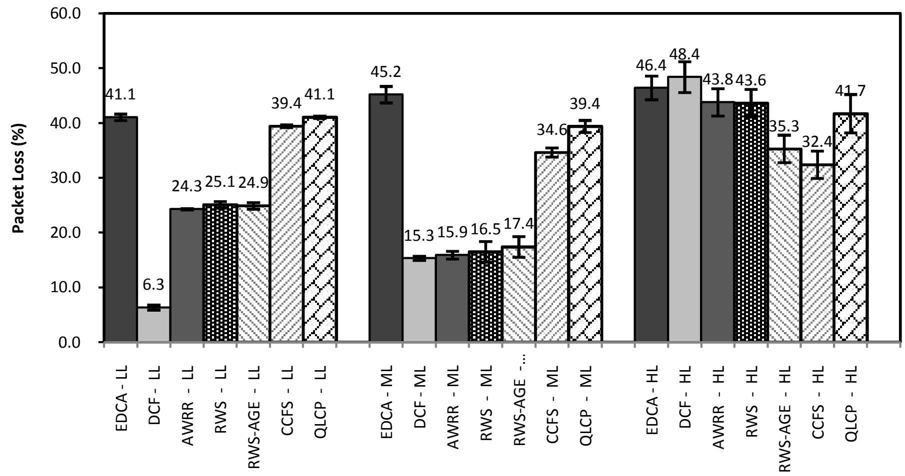
		HP (%)	**MP (%)**	**LP (%)**	**Average (%)**
Low Load	EDCA	41.1	29.9	5.9	25.6
	DCF	6.3	6.3	6.3	6.3
	AWRR	24.3	22.7	5.8	17.6
	RWS	25.1	23.8	6.2	18.4
	RWS-AGE	24.9	23.8	6.0	18.2
	CCFS	39.4	24.8	5.8	23.3
	QLCP	41.1	25.7	6.6	24.5
Medium Load	EDCA	45.2	35.0	14.7	31.6
	DCF	15.3	15.8	15.5	15.5
	AWRR	15.9	15.7	8.5	13.4
	RWS	16.5	16.7	9.5	14.2
	RWS-AGE	17.4	17.5	11.7	15.5
	CCFS	34.6	25.3	10.9	23.6
	QLCP	39.4	27.2	10.4	25.7
High Load	EDCA	46.4	39.8	57.9	48.0
	DCF	48.4	48.4	48.8	48.5
	AWRR	43.8	36.7	35.9	38.8
	RWS	43.6	37.1	34.4	38.4
	RWS-AGE	35.3	38.7	35.1	36.4
	CCFS	32.4	33.8	59.0	41.7
	QLCP	41.7	42.6	40.5	41.6

Table 4.5: Packet Loss in Topology 2.

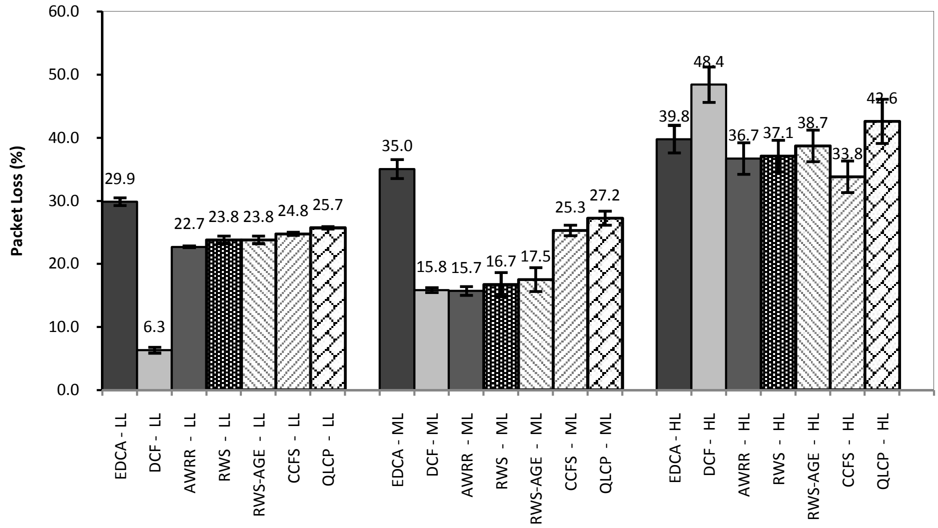
		HP (%)	**MP (%)**	**LP (%)**	**Average (%)**
Low Load	EDCA	46.6	25.6	7.8	26.7
	DCF	7.8	7.9	7.8	7.8
	AWRR	22.4	17.8	5.6	15.3
	RWS	22.3	20.8	7.6	16.9
	RWS-AGE	22.6	19.6	6.6	16.3
	CCFS	38.3	16.1	6.8	20.4
	QLCP	43.1	16.2	6.4	21.9
Medium Load	EDCA	49.3	31.8	17.9	33.0
	DCF	18.4	19.1	18.6	18.7
	AWRR	19.4	18.9	10.9	16.4
	RWS	19.4	18.5	11.3	16.4
	RWS-AGE	22.3	21.4	13.5	19.1
	CCFS	37.6	21.5	12.7	23.9
	QLCP	46.2	25.4	13.8	28.5
High Load	EDCA	60.2	52.7	76.1	63.0
	DCF	61.3	61.5	61.5	61.4
	AWRR	60.2	49.6	46.3	52.0
	RWS	59.5	50.8	46.4	52.2
	RWS-AGE	46.6	51.4	46.7	48.2
	CCFS	43.6	43.1	74.8	53.9
	QLCP	54.3	53.2	50.7	52.7

Table 4.6: Packet Loss in Topology 3.

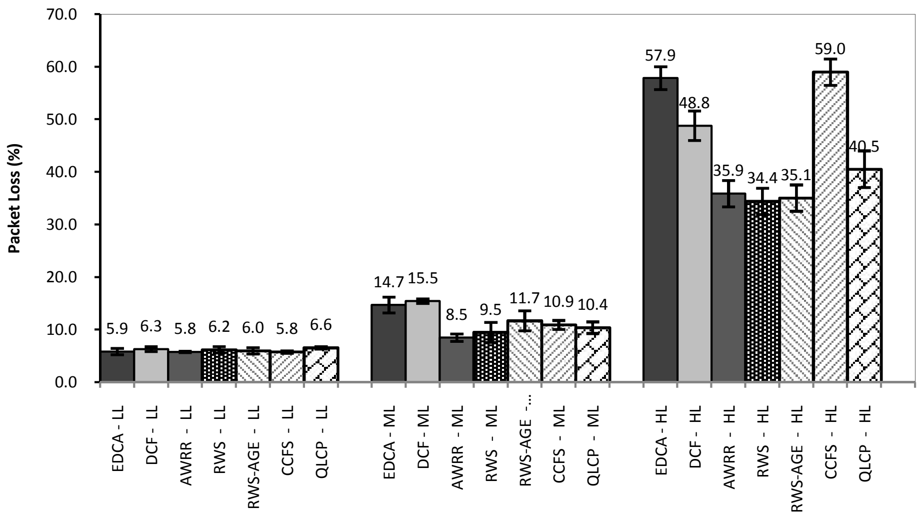
		HP (%)	**MP (%)**	**LP (%)**	**Average (%)**
Low Load	EDCA	25.1	16.0	9.3	16.8
	DCF	6.9	6.9	6.9	6.9
	AWRR	7.9	7.3	6.7	7.3
	RWS	8.5	7.9	7.4	7.9
	RWS-AGE	8.0	7.6	6.7	7.4
	CCFS	14.3	9.7	7.3	10.5
	QLCP	19.6	13.0	8.8	13.8
Medium Load	EDCA	41.3	34.8	31.5	35.9
	DCF	20.1	20.1	20.1	20.1
	AWRR	14.1	13.4	12.8	13.4
	RWS	15.0	14.4	13.8	14.4
	RWS-AGE	15.4	15.2	14.4	15.0
	CCFS	25.0	21.2	19.0	21.7
	QLCP	27.3	21.3	17.5	22.0
High Load	EDCA	54.3	49.8	56.1	53.4
	DCF	56.1	56.4	57.1	56.5
	AWRR	37.7	34.4	32.8	35.0
	RWS	31.8	28.4	26.4	28.9
	RWS-AGE	32.9	33.3	30.8	32.3
	CCFS	42.3	41.1	40.2	41.2
	QLCP	36.7	38.4	36.7	37.3



(a) High Priority Data



(b) Medium Priority Data



(c) Low Priority Data

Figure 4.5: Packet loss with the different scheduling strategies in Topology 1.

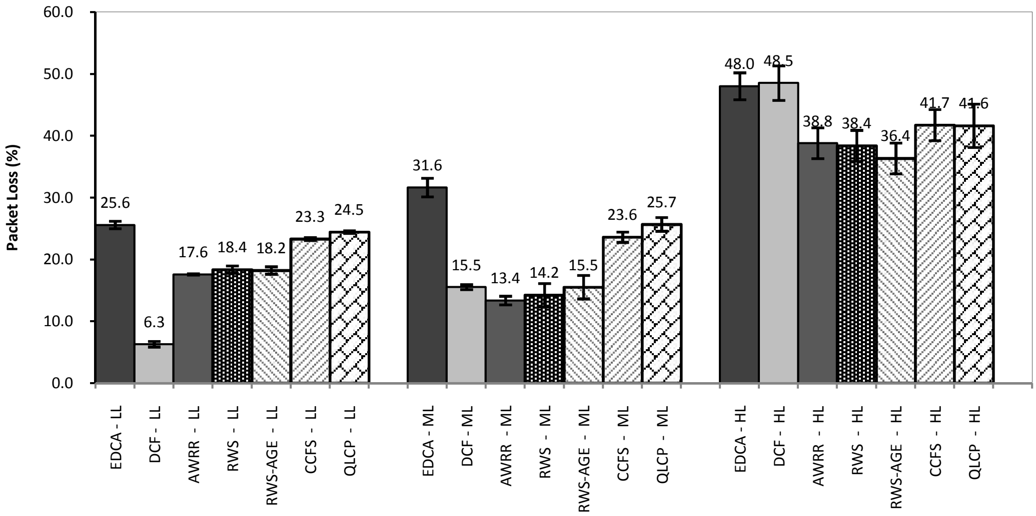


Figure 4.6: Average Packet loss with the different scheduling strategies in Topology 1.

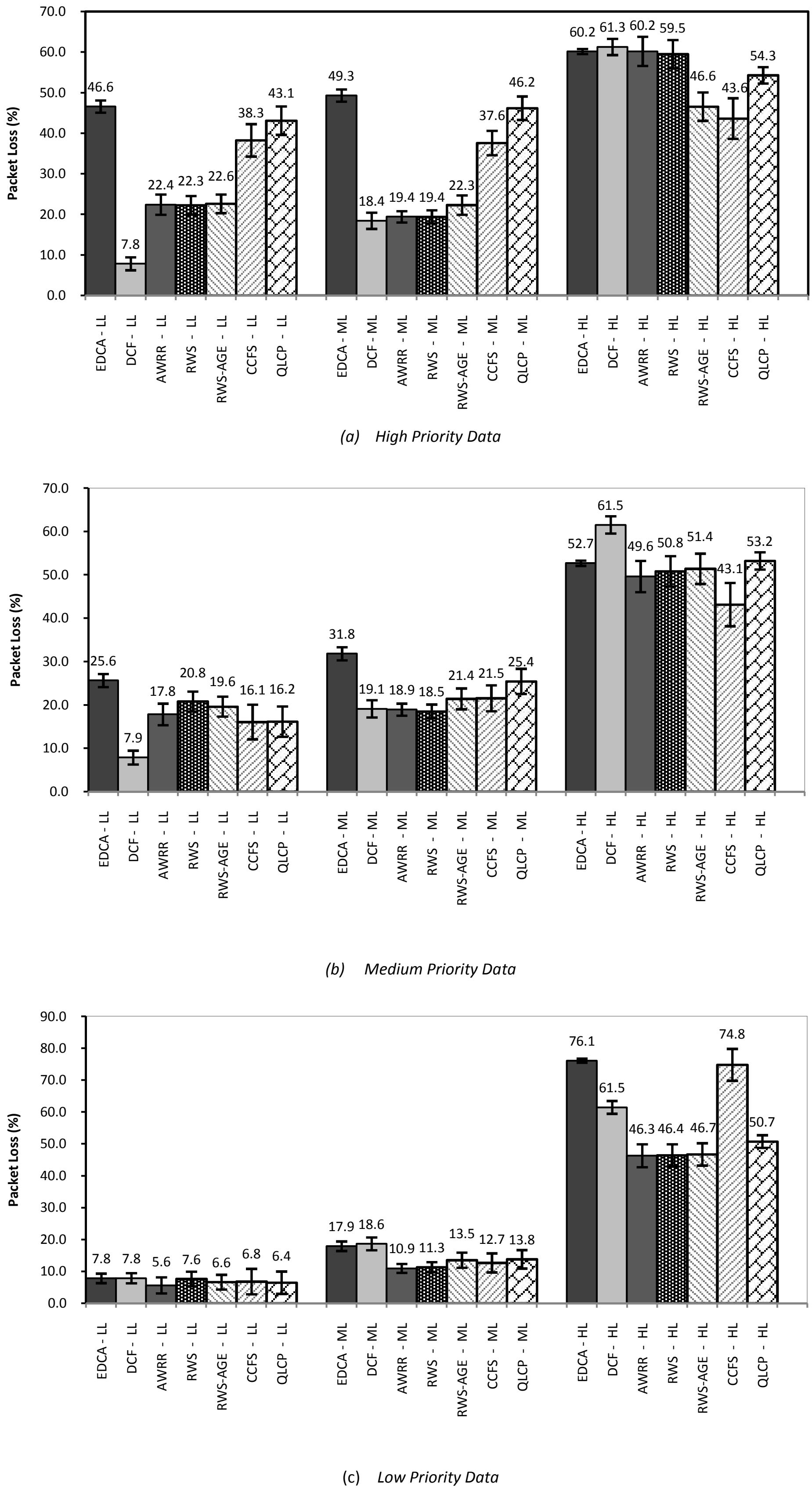


Figure 4.7: Packet loss with the different scheduling strategies in Topology 2.

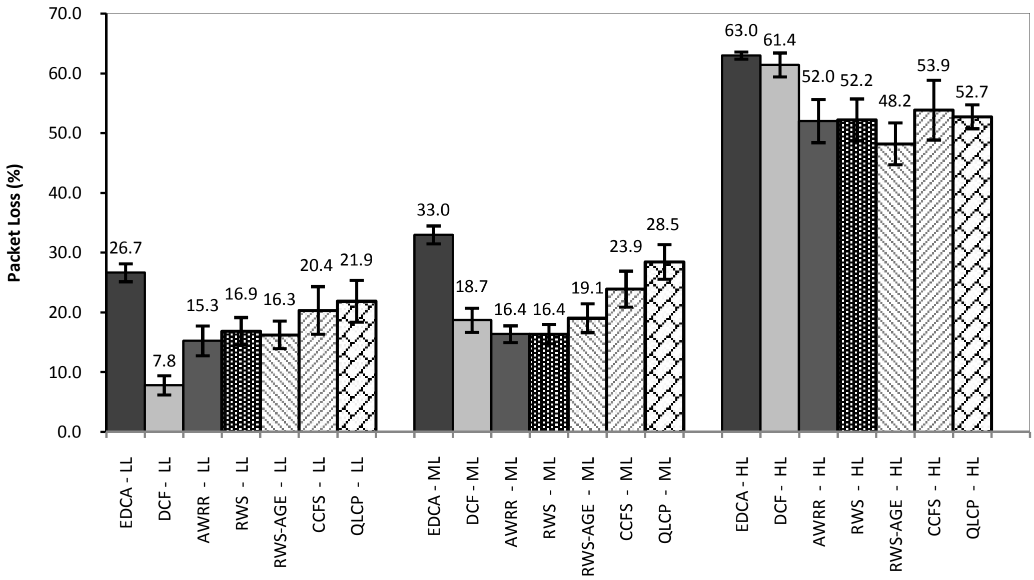
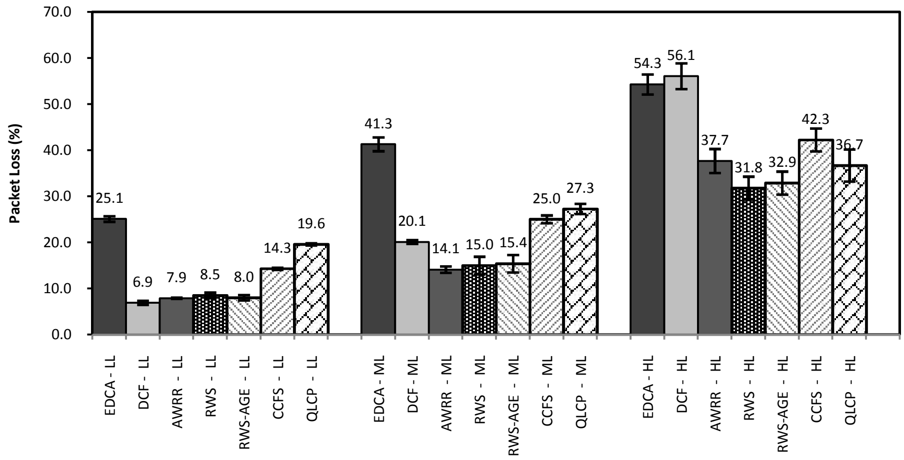
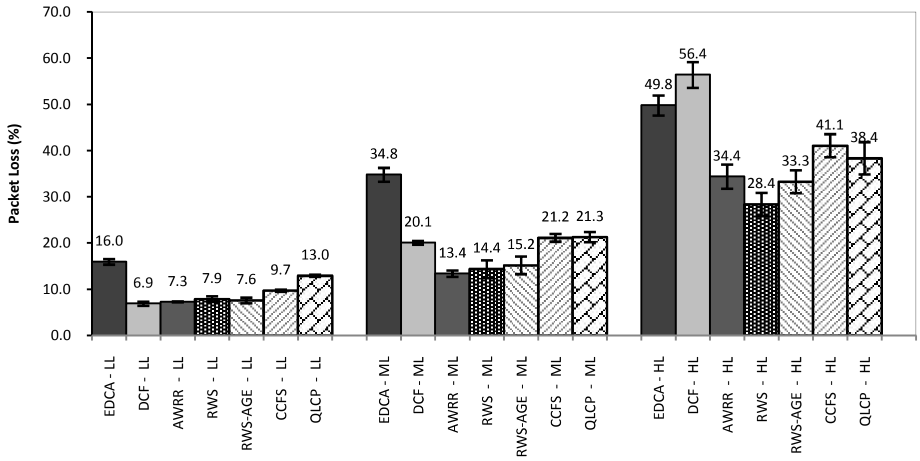


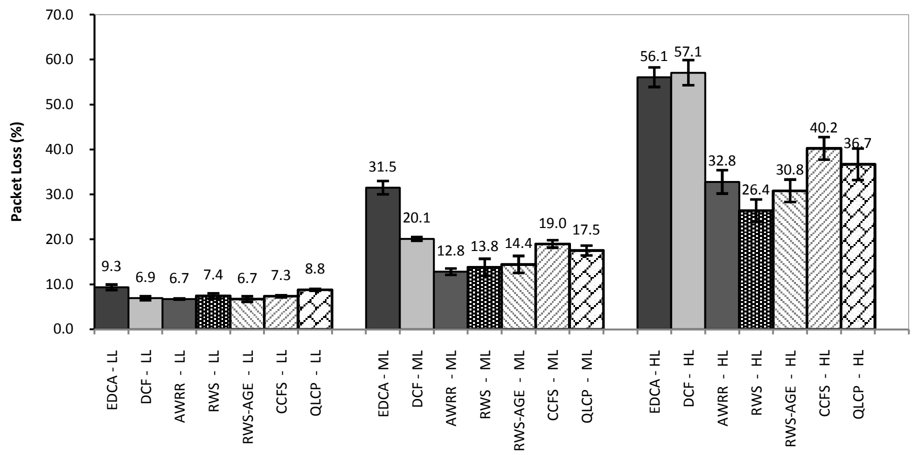
Figure 4.8: Average Packet loss with the different scheduling strategies in Topology 2.



(a) High Priority Data



(b) Medium Priority Data



(c) Low Priority Data

Figure 4.9: Packet loss with the different scheduling strategies in Topology 3.

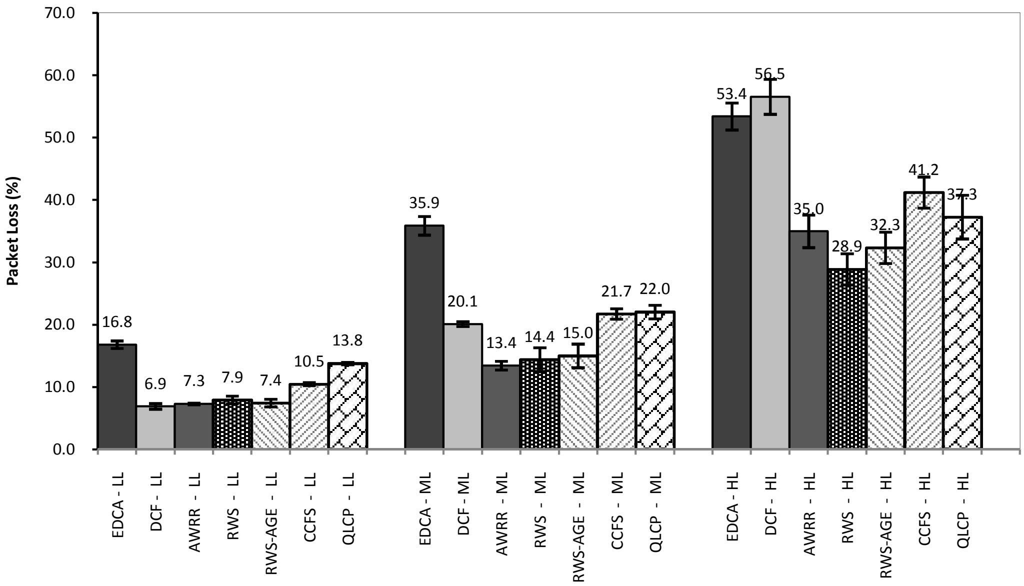


Figure 4.10: Average Packet loss with the different scheduling strategies in Topology 3.

4.8.1.3 End-to-End Delay

Table 4.7 presents a summary of the end-to-end delay for all the different priority data classes over the different network loads in topology 1. Table 4.8 presents a summary of the end-to-end delay in topology 2 and table 4.9 presents a summary of the end-to-end delay in topology 3. Figure 4.11 presents the end-to-end delay experienced with the different priority data in topology 1 and the average calculated end-to-end delay experienced in topology 1 is presented in figure 4.12. Figure 4.13 presents the end-to-end delay experienced with the different priority data in topology 2 and the average calculated end-to-end delay experienced in topology 2 is presented in figure 4.14. Figure 4.15 presents the end-to-end delay experienced with the different priority data in topology 3 and the average calculated end-to-end delay experienced in topology 3 is presented in figure 4.16.

The lowest average end-to-end delay is experienced for high and medium priority data with the use of EDCA as the higher priority data packets have smallest back-off time periods; and overall the higher priority data have a higher probability to access the channel. With DCF all the data priority have the same DIFS and back-off time periods (back-off time depends on the random number selected). For HP and MP data, DCF experienced the longest end-to-end delay time period.

For HP and MP data under all load levels, the AWRR, RWS, RWS-AGE, CCFS and QLCP scheduling mechanisms experience lower end-to-end delay than DCF but more than EDCA. For LP data under all load levels, the AWRR, RWS, RWS-AGE and QLCP scheduling mechanisms experience lower end-to-end delay than DCF and EDCA, while CCFS experiences more delay than EDCA.

With DCF for HP, MP and LP, roughly the same average end-to-end delay is experienced as the packets are all treated with equal priority in a FIFO fashion. It is observed that with the RWS and AWRR scheduling mechanisms, LP data experiences a lower end-to-end delay compared to HP and MP data. This is as a result of the increase in the chances of lower priority data gaining access to the channel and transmitting their data. With the RWS-AGE mechanism, lower end-to-end delay is experienced for HP compared to MP and LP data in all the test topologies and load levels. This happens as a result of transmission from another priority queue in the event that the same priority queue packet is transmitted consecutively lowering the changes of collision when lower priority data is transmitted.

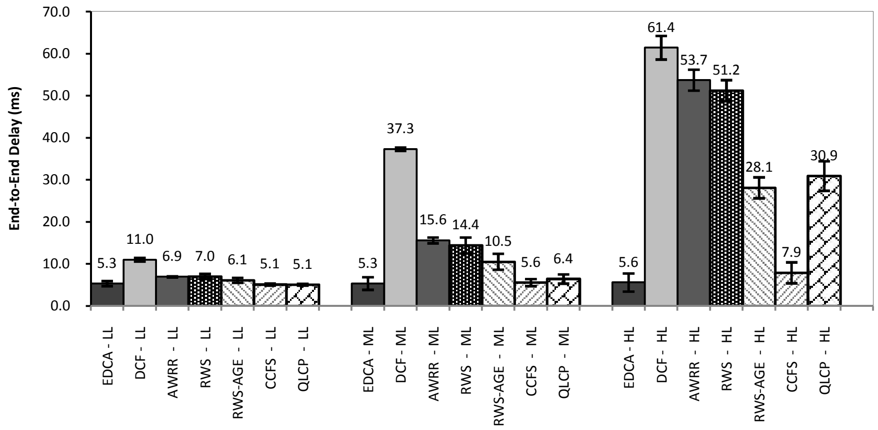
The number of collisions has a significant effect on the achievable end-to-end delay as well. Overall between AWRR, RWS and RWS-AGE, RWS-AGE experienced lower end-to-end delay for HP data. The RWS-AGE mechanism experienced the lowest end-to-end delay for HP data, while for MP and LP, it experienced higher end-to-end delay. This provides the answer to our research question on which SBC mechanism on average results in the lowest end-to-end delay. The application of RWS-AGE on HP data results in the least end-to-end delay out of all the SBC mechanisms investigated with a 95% confidence level except with CCFS which starves LP data.

Table 4.7: Summary of End-to-End Delay in Topology 1.

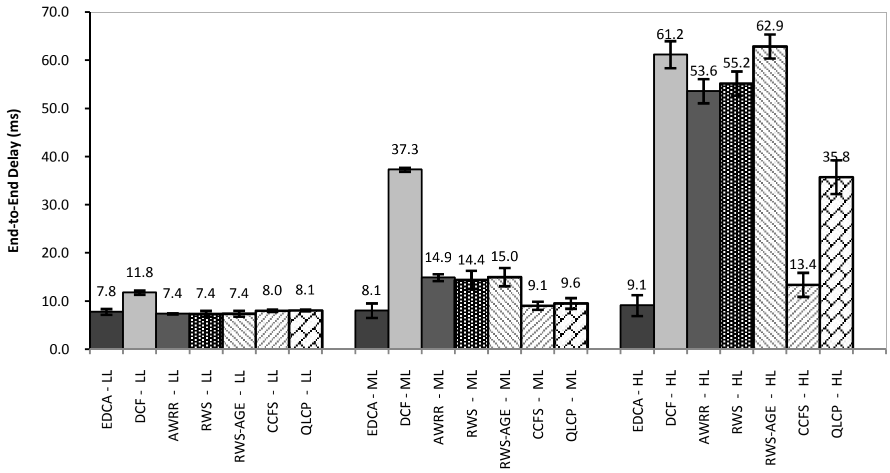
		HP (ms)	**MP (ms)**	**LP (ms)**	**Average (ms)**
Low Load	EDCA	5.3	7.8	5.9	6.3
	DCF	11.0	11.8	11.7	11.5
	AWRR	6.9	7.4	5.9	6.7
	RWS	7.0	7.4	5.9	6.8
	RWS-AGE	6.1	7.4	5.8	6.4
	CCFS	5.1	8.0	5.9	6.3
	QLCP	5.1	8.1	5.9	6.4
Medium Load	EDCA	5.3	8.1	16.8	10.1
	DCF	37.3	37.3	37.1	37.2
	AWRR	15.6	14.9	13.8	14.8
	RWS	14.4	14.4	13.8	14.2
	RWS-AGE	10.5	15.0	16.2	13.9
	CCFS	5.6	9.1	19.7	11.4
	QLCP	6.4	9.6	19.0	11.7
High Load	EDCA	5.6	9.1	143.2	52.6
	DCF	61.4	61.2	61.4	61.3
	AWRR	53.7	53.6	35.9	47.7
	RWS	51.2	55.2	38.5	48.3
	RWS-AGE	28.1	62.9	58.7	49.9
	CCFS	7.9	13.4	216.6	79.3
	QLCP	30.9	35.8	97.7	54.8

Table 4.8: Summary of End-to-End Delay in Topology 2.

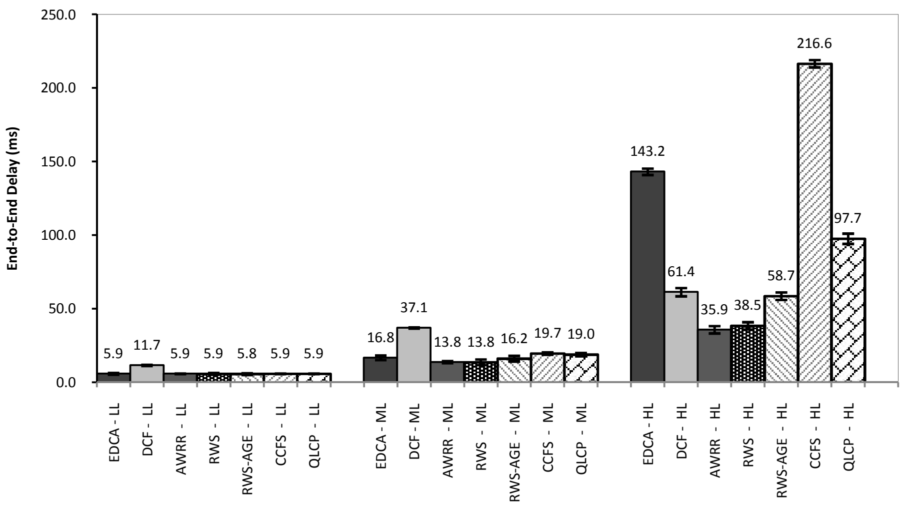
		HP (ms)	**MP (ms)**	**LP (ms)**	**Average (ms)**
Low Load	EDCA	6.7	8.4	6.0	7.0
	DCF	11.6	11.6	11.6	11.6
	AWRR	8.6	8.2	6.0	7.6
	RWS	8.6	8.2	6.0	7.6
	RWS-AGE	7.8	8.4	6.0	7.4
	CCFS	6.6	8.7	6.0	7.1
	QLCP	6.5	8.6	6.1	7.0
Medium Load	EDCA	6.8	8.9	22.0	12.6
	DCF	42.4	42.5	42.5	42.5
	AWRR	19.4	16.1	15.7	17.1
	RWS	15.6	15.4	15.4	15.5
	RWS-AGE	11.6	16.2	18.8	15.5
	CCFS	6.8	9.8	9.7	8.8
	QLCP	8.7	11.0	25.0	14.9
High Load	EDCA	7.5	10.8	181.7	66.7
	DCF	91.5	91.9	91.9	91.8
	AWRR	88.2	84.3	53.6	75.5
	RWS	77.8	79.7	53.7	70.4
	RWS-AGE	33.4	86.6	92.9	71.0
	CCFS	9.6	15.3	313.2	112.7
	QLCP	48.0	53.1	142.0	81.1



(a) High Priority Data



(b) Medium Priority Data



(c) Low Priority Data

Figure 4.11: End-to-End Delay with the different scheduling strategies in Topology 1.

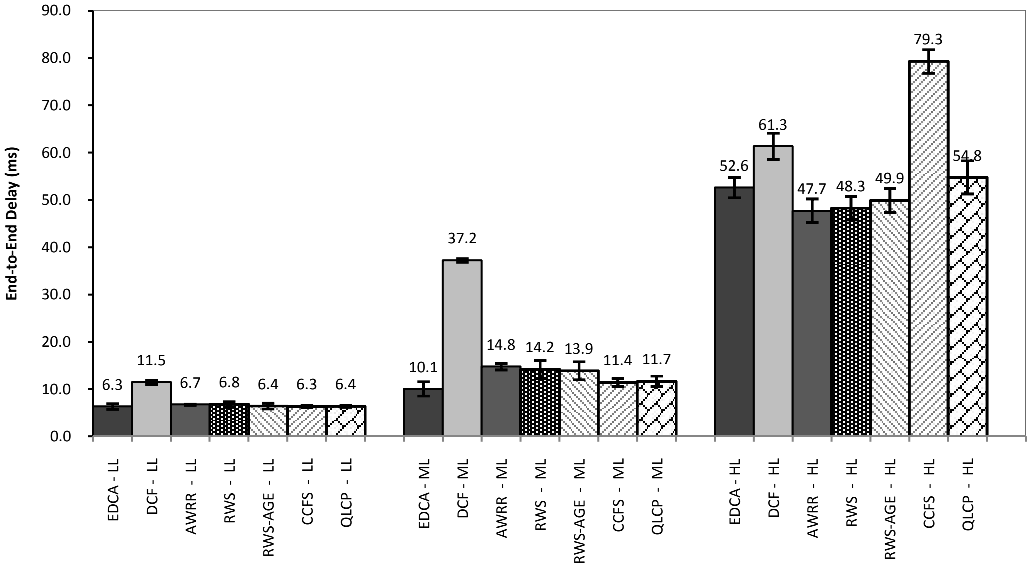
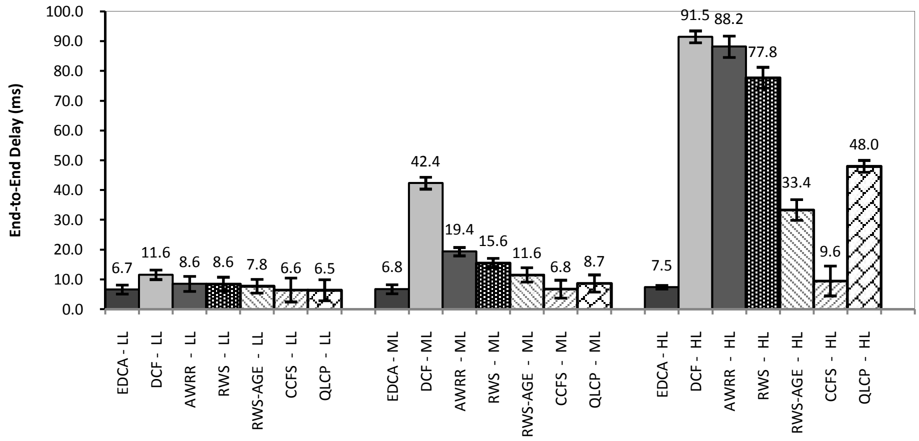


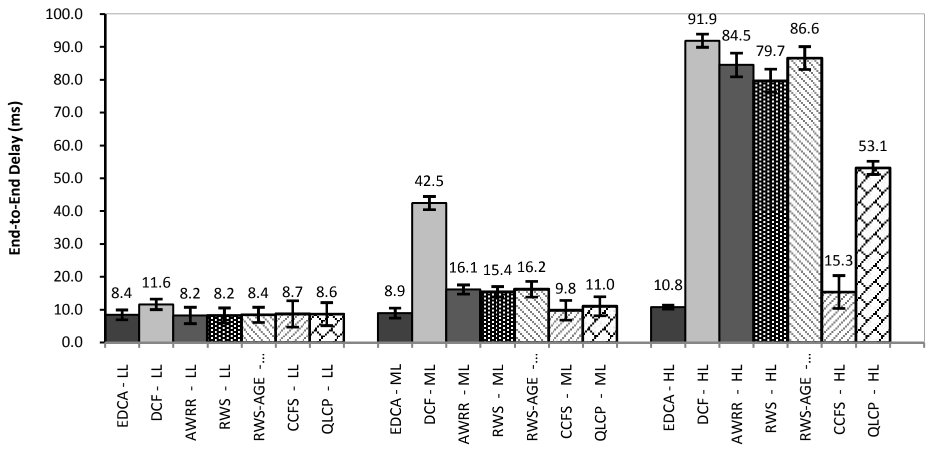
Figure 4.12: Average End-to-End Delay with the different scheduling strategies in Topology 1.

Table 4.9: Summary of End-to-End Delay in Topology 3.

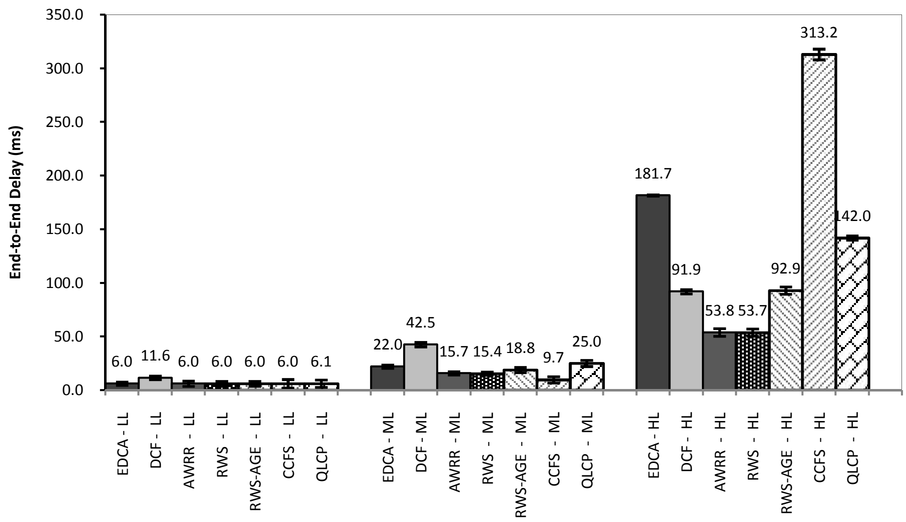
		HP (ms)	MP (ms)	LP (ms)	Average (ms)
Low Load	EDCA	5.5	7.1	15.4	9.3
	DCF	13.4	13.5	14.1	13.7
	AWRR	11.4	11.1	10.2	10.9
	RWS	11.3	11.1	10.2	10.9
	RWS-AGE	9.1	11.1	11.1	10.4
	CCFS	5.8	8.2	12.6	8.9
	QLCP	4.7	7.8	12.4	8.3
Medium Load	EDCA	5.5	7.8	19.3	10.9
	DCF	15.2	15.1	16.8	15.7
	AWRR	11.7	11.4	10.4	11.2
	RWS	11.6	11.3	10.3	11.1
	RWS-AGE	9.3	11.5	11.4	10.7
	CCFS	5.9	8.2	13.4	9.2
	QLCP	4.9	7.9	13.1	8.6
High Load	EDCA	5.6	8.1	67.4	27.0
	DCF	40.8	41.4	42.2	41.5
	AWRR	19.5	18.3	13.9	17.2
	RWS	19.7	18.1	14.2	17.3
	RWS-AGE	13.1	22.8	19.7	18.5
	CCFS	6.8	9.3	61.6	25.9
	QLCP	11.5	14.7	39.1	21.8



(a) High Priority Data



(b) Medium Priority Data



(c) Low Priority Data

Figure 4.13: End-to-End Delay with the different scheduling strategies in Topology 2.

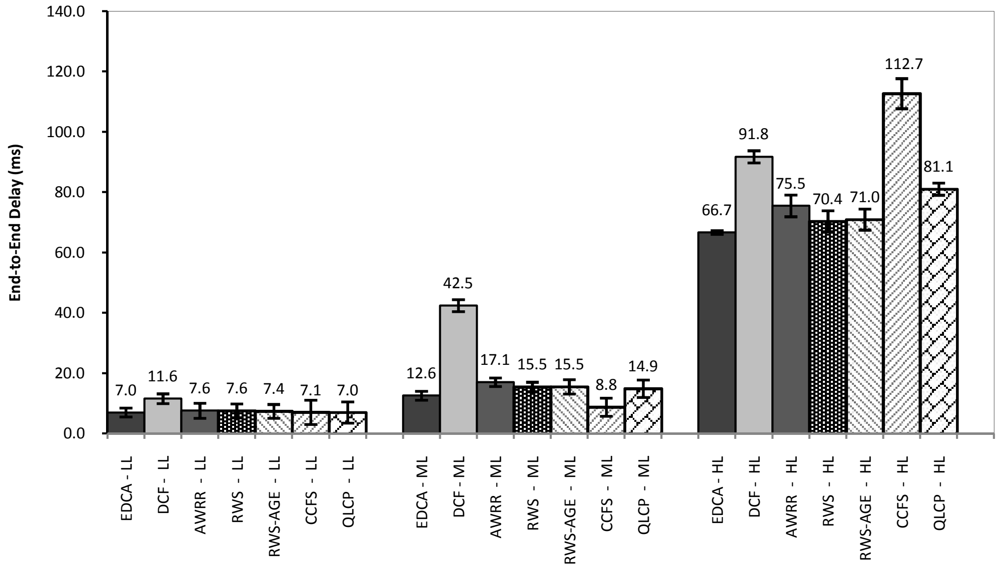
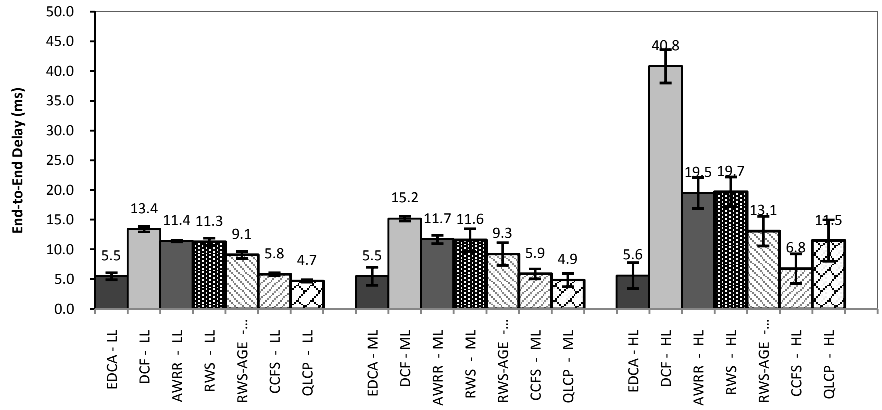
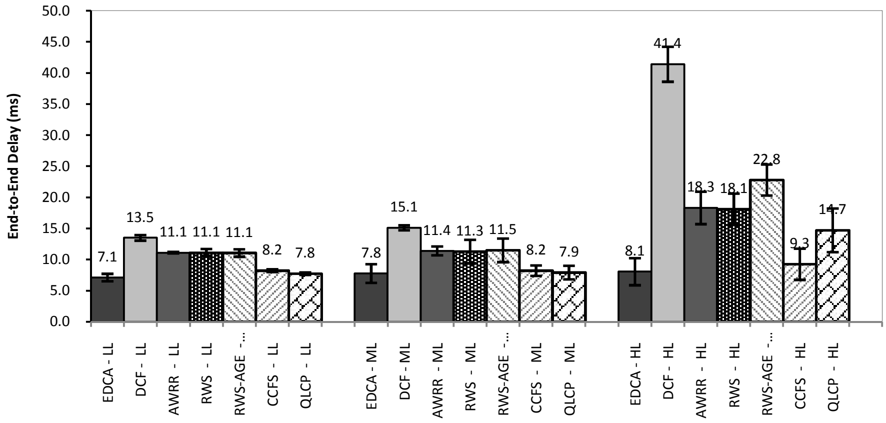


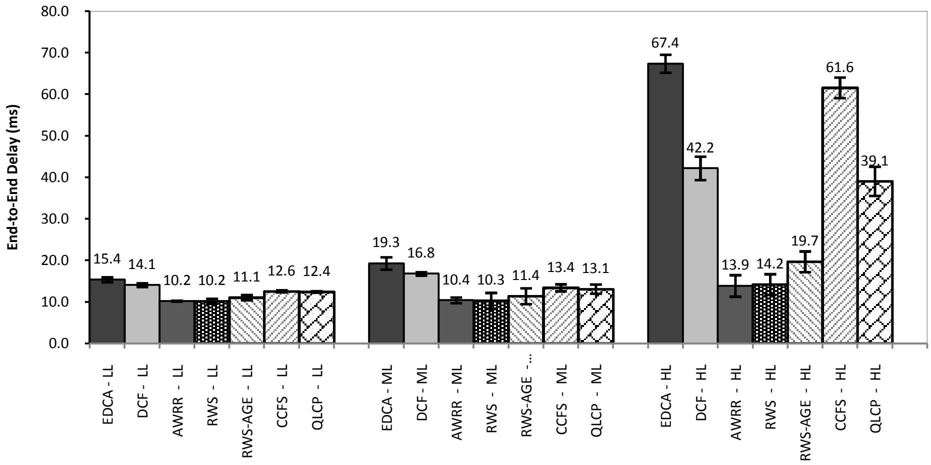
Figure 4.14: Average End-to-End Delay with the different scheduling strategies in Topology 2.



(a) High Priority Data



(b) Medium Priority Data



(c) Low Priority Data

Figure 4.15: End-to-End Delay with the different scheduling strategies in Topology 3.

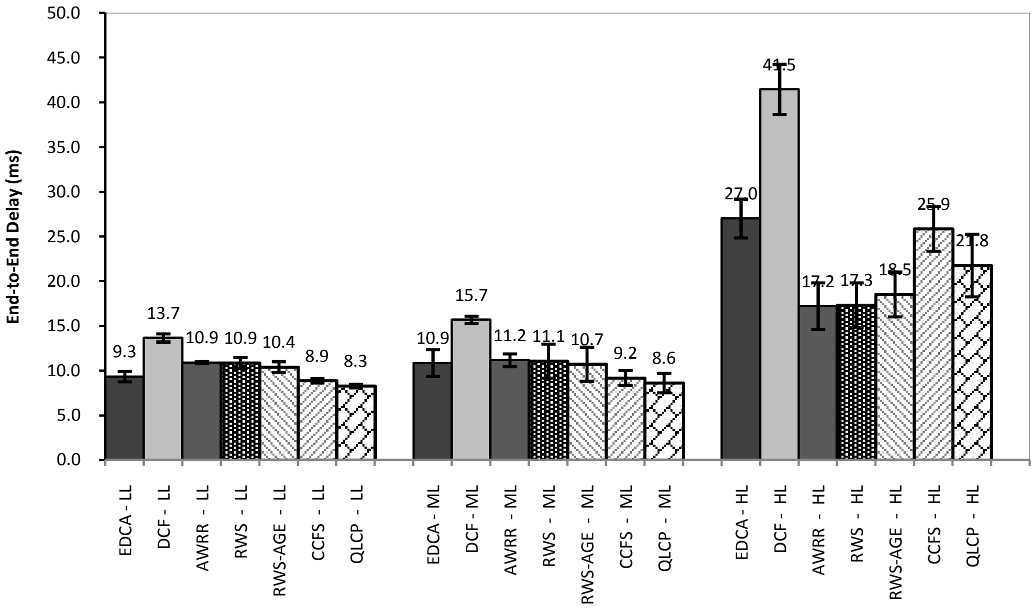


Figure 4.16: Average End-to-End Delay with the different scheduling strategies in Topology 3.

4.8.1.4 Jain's Fairness Index

Figures 4.17, 4.18 and 4.19 present the Jain's fairness index for the heavy load scenarios for the test topologies 1, 2 and 3 with the different scheduling strategies. It can be noted that for all the test topologies, EDCA provided the least fairness and DCF provided the highest fairness. With AWRR, RWS, RWS-AGE and QLCP scheduling strategies, fairness is considerably improved compared to EDCA. However, the JFI is less than DCF. CCFS has a lower fairness compared to EDCA, DCF, AWRR, RWS, RWS-AGE and QLCP in topologies 1 and 2. The results show that the CCFS mechanism starves lower priority data under heavy loads and has the lowest JFI values. The more a scheme starves the lower priority packets, the lower will be the JFI value. Higher JFI values indicate a higher chance of equal number of packets for each priority to successfully reach the destination.

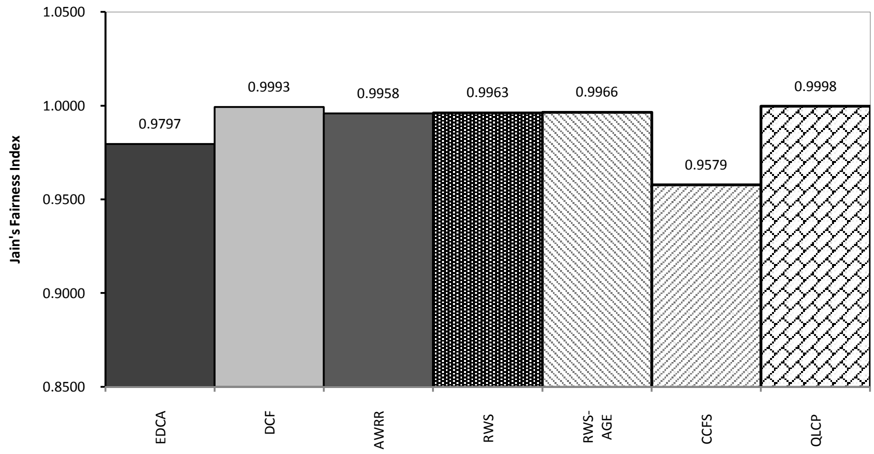


Figure 4.17: Fairness measure under heavy load with the different scheduling strategies in Topology 1.

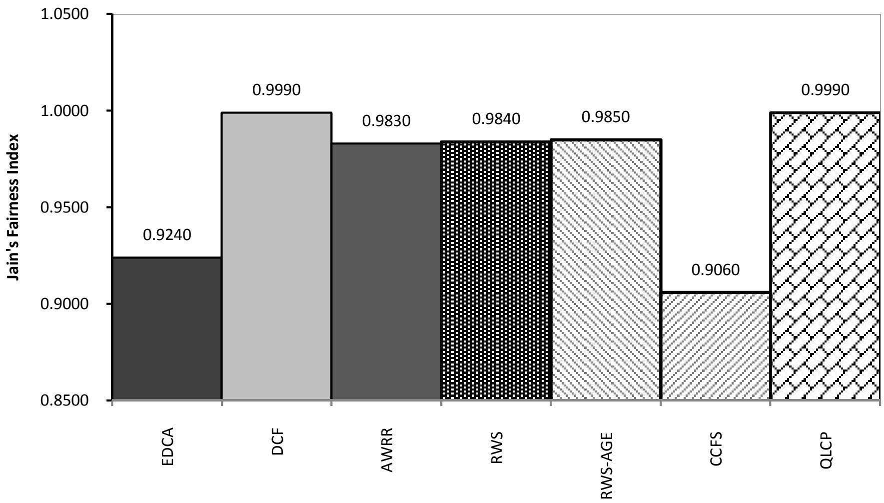


Figure 4.18: Fairness measure under heavy load with the different scheduling strategies in Topology 2.

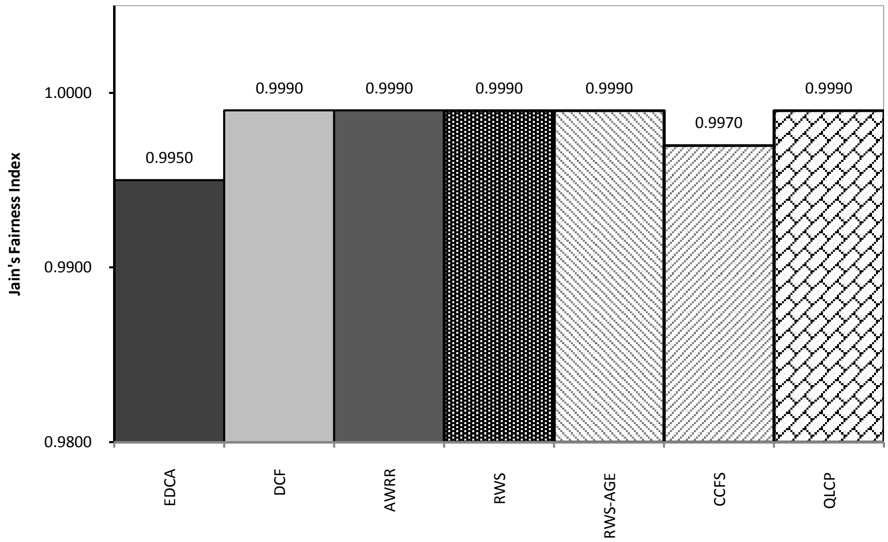


Figure 4.19: Fairness measure under heavy load with the different scheduling strategies in Topology 3.

4.8.2 Performance of the strategies with the use of TXOP

In this section, we apply TXOP to the scheduling strategies and present the results. The values of TXOP are chosen such that more higher priority packets must be allowed to be transmitted compared to the lower priority packets to provide higher priority as well as since higher priority packets having smaller CW sizes result in more collisions. Values of 2 *ms* for HP, 1 *ms* for MP and 0 for LP were used to obtain the results in this section as to determine if TXOP improves performance in SBC strategies. DCF only has one queue and no TXOP is applied to DCF. The results for DCF presented in this section are without the application of TXOP.

4.8.2.1 Packet Loss

Table 4.10 presents a summary of the packet loss for all the different priority data classes over the different network loads in topology 1 with TXOP. Table 4.11 presents a summary of the packet loss in topology 2 with TXOP. Figure 4.20 presents the average calculated packet loss experienced in topology 1, while figure 4.21 presents the average calculated packet loss in topology 2 with TXOP. A packet loss reduction of 1.6% for HP data, 1.2% for MP and 12.0% for LP in EDCA can be observed with the application of TXOP in topology 1 under high loads. A packet loss reduction of 2.7% for LP in AWRR can be observed with the application of TXOP in topology 2. A packet loss reduction of 6% for HP data, 5.9% for MP and 4% for LP in RWS can be observed with the application of TXOP in topology 1 under high loads. A packet loss reduction of 3.1% for HP data, 3.1% for MP and 14.2% for LP in CCFS can be observed with the application of TXOP in topology 1. A packet loss reduction of 4.4% for HP data, 5.9% for MP and 34.7% for LP in QLCP can be observed with the application of TXOP in topology 1 under high loads. Higher packet loss reduction for HP and MP data is observed with the mechanisms that do not starve lower priority data such as AWRR, RWS, RWS-AGE and QLCP.

The results provide the answer to our research question on whether the use of TXOP in SBC strategies improve performance in terms of lowering packet loss. The use of TXOP helps lower packet loss. With the application of TXOP to EDCA, a significant reduction of packet loss for HP is not observed as already HP data are gaining access to the medium more frequently under heavy loads compared to the LP data. TXOP with EDCA does result in significant packet loss reduction for LP data of up to 12%. A 14.2% packet loss reduction for LP with CCFS is also observed. CCFS with TXOP performs better than EDCA on average in terms of less packet loss.

Table 4.10: Packet Loss in Topology 1 with TXOP.

		HP (%)	**MP (%)**	**LP (%)**	**Average (%)**
Low Load	EDCA	41.4	29.4	6.1	25.5
	DCF	6.3	6.3	6.3	6.3
	AWRR	24.1	23.1	6.2	17.4
	RWS	23.9	22.7	5.8	17.4
	RWS-AGE	24.5	23.4	6.3	18.1
	CCFS	39.4	25.0	5.8	23.4
	QLCP	39.5	26.3	5.9	25.0
Medium Load	EDCA	42.3	32.4	11.0	28.6
	DCF	15.3	15.8	15.5	15.5
	AWRR	16.6	17.0	9.3	13.8
	RWS	17.1	17.2	9.45	15.3
	RWS-AGE	17.0	17.9	9.25	13.7
	CCFS	34.7	24.7	9.0	22.8
	QLCP	39.2	26.2	29.8	31.7
High Load	EDCA	44.8	38.6	45.9	43.1
	DCF	48.4	48.4	48.8	48.5
	AWRR	35.2	29.7	27.6	30.8
	RWS	37.6	31.2	30.46	33.1
	RWS-AGE	32.5	33.6	30.2	32.1
	CCFS	29.3	30.7	44.9	35.0
	QLCP	37.3	36.7	29.8	34.6

Table 4.11: Packet Loss in Topology 2 with TXOP.

		HP (%)	**MP (%)**	**LP (%)**	**Average (%)**
Low Load	EDCA	45.7	24.8	6.4	25.6
	DCF	7.8	7.9	7.8	7.8
	AWRR	22.6	19.4	6.8	17.1
	RWS	21.9	18.4	6.5	15.6
	RWS-AGE	22.4	20.1	5.8	16.1
	CCFS	37.6	16.0	6.0	19.9
	QLCP	41.5	14.8	6.2	20.9
Medium Load	EDCA	48.9	30.3	14.6	31.2
	DCF	18.4	19.1	18.6	18.7
	AWRR	18.4	17.5	9.8	17.6
	RWS	18.2	17.3	10.1	14.6
	RWS-AGE	18.6	18.1	9.8	15.5
	CCFS	36.7	19.2	8.6	21.5
	QLCP	45.8	23.9	12.4	27.4
High Load	EDCA	57.5	49.8	65.1	57.4
	DCF	61.3	61.5	61.5	61.4
	AWRR	48.4	40.5	36.64	41.8
	RWS	48.7	41.8	37.4	42.6
	RWS-AGE	42.5	44.1	39.6	42.1
	CCFS	44.5	43.7	68.1	52.1
	QLCP	52.7	49.9	43.3	48.6

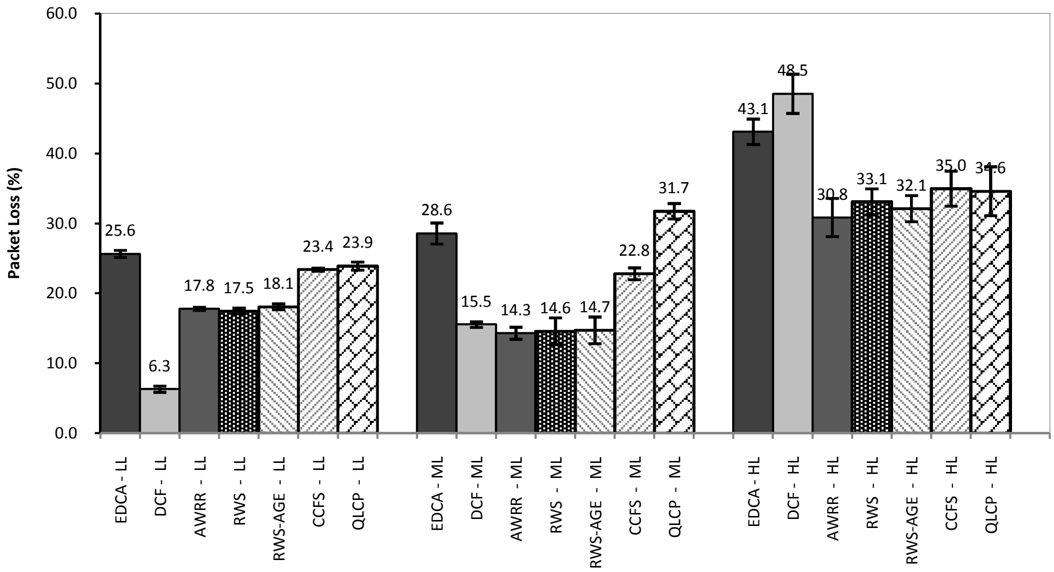


Figure 4.20: Average Packet loss with the different scheduling strategies in Topology 1 with TXOP.

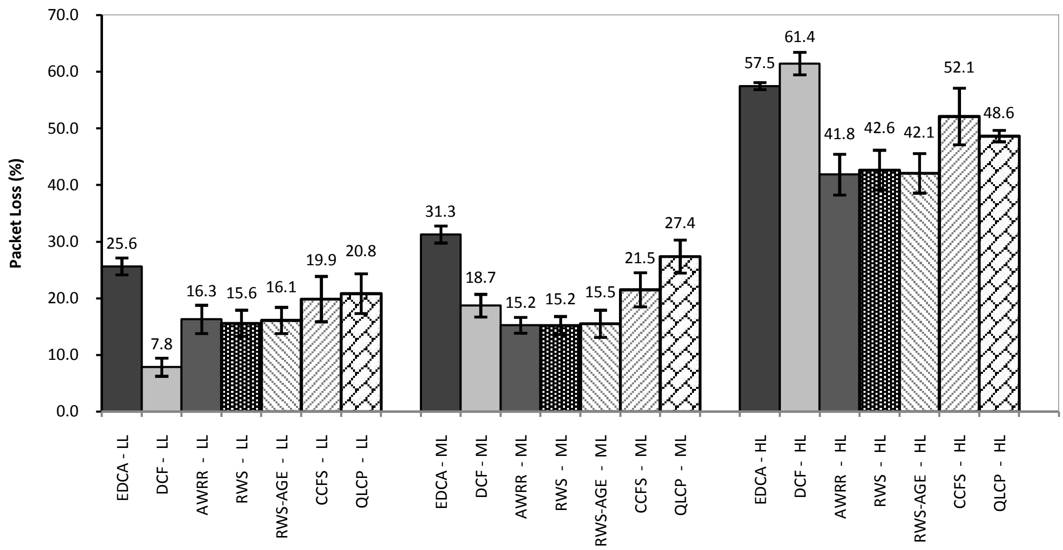


Figure 4.21: Average Packet loss with the different scheduling strategies in Topology 2 with TXOP.

Table 4.12: Summary of End-to-End Delay in Topology 1 with TXOP.

		HP (ms)	**MP (ms)**	**LP (ms)**	**Average (ms)**
Low Load	EDCA	5.3	7.8	5.9	6.3
	DCF	11.0	11.8	11.7	11.5
	AWRR	6.3	7.3	5.9	6.5
	RWS	7.0	7.4	5.9	6.7
	RWS-AGE	6.2	7.4	5.9	6.5
	CCFS	5.2	8.1	5.9	6.4
	QLCP	5.1	7.6	5.9	6.2
Medium Load	EDCA	5.3	8.0	15.6	9.6
	DCF	37.3	37.3	37.1	37.2
	AWRR	14.3	14.0	13.5	13.9
	RWS	13.7	13.6	13.6	13.6
	RWS-AGE	9.9	13.4	14.8	12.7
	CCFS	5.6	9.1	17.7	10.8
	QLCP	6.0	9.1	16.3	10.5
High Load	EDCA	5.3	8.8	86.3	33.5
	DCF	61.4	61.2	61.4	61.3
	AWRR	46.7	46.2	34.7	42.5
	RWS	44.3	46.1	36.5	42.3
	RWS-AGE	26.7	53.9	51.2	43.9
	CCFS	7.7	13.5	138.9	53.4
	QLCP	24.7	29.1	75.1	43.0

4.8.2.2 End-to-End Delay

Table 4.12 presents a summary of the end-to-end delay for all the different priority data classes over the different network loads in topology 1 with TXOP. Table 4.13 presents a summary of the end-to-end delay in topology 2 with TXOP. Figure 4.22 presents the average calculated end-to-end delay experienced in topology 1, while figure 4.23 presents the average calculated end-to-end delay in topology 2 with TXOP. An end-to-end delay reduction of 0.3 *ms* for HP data, 0.3 *ms* for MP and 56.9 *ms* for LP in EDCA can be observed with the application of TXOP in topology 1 under high loads. An end-to-end delay reduction of 11.8 *ms* for HP data, 9.1 *ms* for MP and 9.7 *ms* for LP in AWRR can be observed with the application of TXOP in topology 2. An end-to-end delay reduction of 1.4 for HP data, 9.0 *ms* for MP and 7.5 *ms* for LP in RWS-AGE can be observed with the application of TXOP in topology 1 under high loads. An end-to-end delay reduction of 0.2 *ms* for HP data and 77.7 *ms* for LP in CCFS can be observed with the application of TXOP in topology 1. Higher end-to-end delay reduction for HP and MP data is also observed with the SBC mechanisms that do not starve lower priority data. The application of TXOP is an ideal way to lower the end-to-end delay in SBC strategies.

These results also provide answers to our research question on whether the use of TXOP in SBC strategies improve performance in terms of lowering end-to-end delay. The end-to-end delay can be reduced with the application of TXOP. TXOP is an important ingredient in SBC strategies for lowering both packet loss and end-to-end delay.

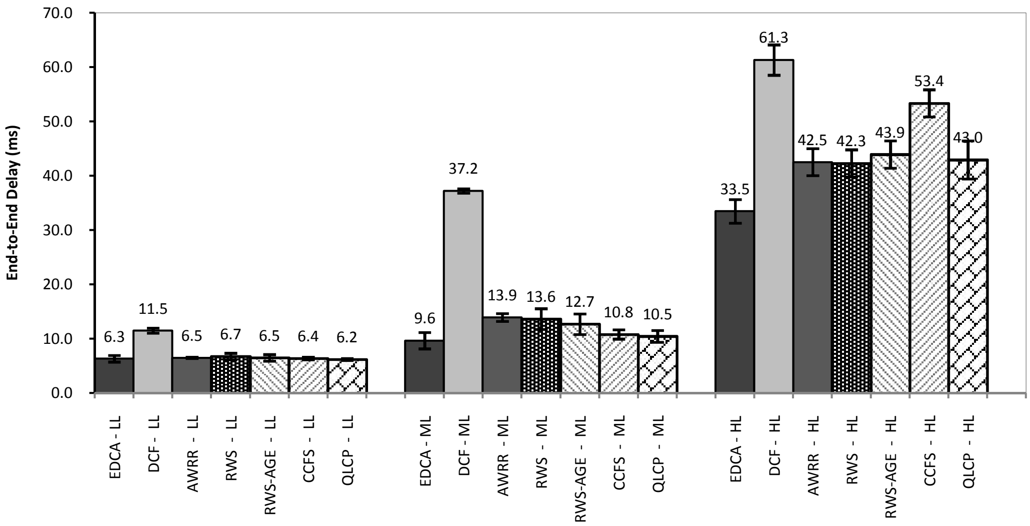


Figure 4.22: Average End-to-End Delay with the different scheduling strategies in Topology 1 with TXOP.

Table 4.13: Summary of End-to-End Delay in Topology 2 with TXOP.

		HP (ms)	MP (ms)	LP (ms)	Average (ms)
Low Load	EDCA	6.7	8.4	5.9	7.0
	DCF	11.6	11.6	11.6	11.6
	AWRR	8.6	8.2	5.9	7.6
	RWS	8.6	8.2	6	7.6
	RWS-AGE	8.6	8.2	6	7.6
	CCFS	6.6	8.7	5.8	7.0
	QLCP	6.4	8.6	5.9	7.0
Medium Load	EDCA	6.8	8.8	19.3	11.6
	DCF	42.4	42.5	42.5	42.5
	AWRR	15.3	14.7	14.9	15.0
	RWS	14.3	13.8	14.5	14.2
	RWS-AGE	11.1	14.4	17.1	14.2
	CCFS	6.8	9.8	21.3	12.6
	QLCP	7.6	10.0	19.5	12.3
High Load	EDCA	7.5	11.1	133.7	50.8
	DCF	91.5	91.9	91.9	91.8
	AWRR	67.8	64.2	46.9	59.6
	RWS	62.5	62.9	48.4	57.9
	RWS-AGE	42.5	44.1	39.7	42.1
	CCFS	10.3	16.9	236.1	87.8
	QLCP	38.2	45.5	106.8	63.5

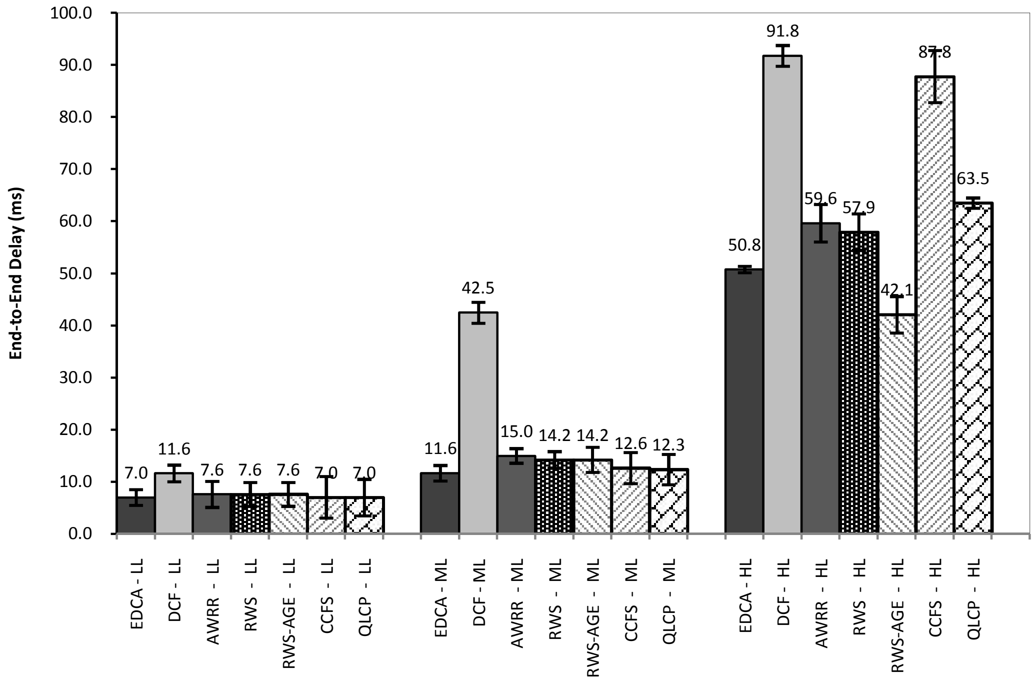


Figure 4.23: Average End-to-End Delay with the different scheduling strategies in Topology 2 with TXOP.

4.9 Conclusion

From our results on the application of DCF to multi-hop networks, we observe that DCF provides a higher degree of reliability in terms of less packet loss compared to EDCA on average. Moreover, DCF provides a very high degree of fairness compared to EDCA as it does not provide data differentiation. However, DCF results in high end-to-end delay compared to EDCA for high and medium priority data.

The AWRR, RWS and RWS-AGE SBC mechanisms have shown to provide lower packet loss than EDCA and DCF in high load scenarios. Overall, the RWS-AGE scheduling SBC mechanism is seen as the best performing scheduling strategy in the homogeneous configured network layout schemes in high load and high contention networks. The criteria used for judgment is to choose a strategy with the lowest packet loss under high loads and on on average, fewer collisions on average and provide low end-to-end delay. The end-to-end delay does not necessarily have to be the lowest, but must be within the tolerable limit range for the application. The RWS-AGE mechanism experiences on average lower packet loss and lower end-to-end delay for HP data under heavy load scenarios. RWS-AGE reduces collisions on the medium and also increases the probability chances of medium and low priority data to access the medium compared to EDCA. The strategy also avoids the internal collision mechanism present in EDCA. Compared to EDCA, overall the lower priority data have a higher chance of transmission with RWS-AGE. In EDCA under heavy loads if all the priority queues have data for transmission, the transmission probability for the lower priority data keeps reducing, while it stays constant with RWS-AGE. Overall in the network, a higher number of lower priority packets will be in transmission than EDCA which results in lowering the collision probability on the network as low priority data use wider CW ranges.

Results show that the design of the scheduling strategy can have a significant impact on the QoS achievable in WMNs. The CCFS SBC mechanism starved lower priority data while the AWRR, RWS and RWS-AGE mechanisms do not starve the lower priority data and show considerable reduction in packet loss and collisions compared to EDCA under heavy loads. The fairness is also improved under heavy load scenarios compared to the use of EDCA.

The application and use of TXOP in SBC has a significant effect on lowering the packet loss and end-to-end delay for higher priority data compared to the use of TXOP with EDCA. In EDCA, TXOP mainly lowers packet loss for lower priority data and end-to-end delay. The use of TXOP is an important ingredient in SBC strategies for lowering both packet loss and end-to-end delay.

Chapter 5

Hybrid Configured Network Layout Schemes

5.1 Introduction

In the previous chapter, the use of homogeneous configured network layout schemes were investigated for the SBC scheduling strategies. In this chapter we investigate the performance of hybrid configured network layouts. In hybrid configured network layout schemes, different nodes are assigned different scheduling strategies. Homogeneous configured network layouts are proposed where no distinction can made between edge and core nodes in a network layout. In some application networks, however, data from different network domains can reach the backbone network through edge nodes. By domain is meant a section of a network where the nodes are connected and they can communicate directly, or through multi-hop communication, in that domain. Traffic exits that part of the network through an edge node that connects to the backbone mesh network. An illustration of a multi-domain network as used in this work is shown in figure 5.1.

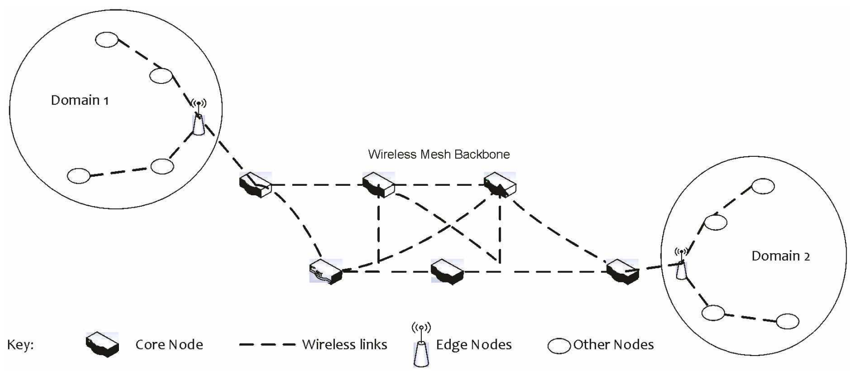


Figure 5.1: A multi-domain network.

This chapter presents the motivation for the experiments, the experimental setup overview, the results and the discussion of the results. In this research, we proposed and investigated many design scheme combinations for the wireless backbone network. A hierarchical backbone mesh network structure, consisting of edge and core routers is considered where user clients can connect to the edge routers and the core routers connect to the backbone routers. All the network design schemes investigated employ a single-radio and a single-channel for both edge and core devices. This will cause congestion and bottlenecks in a network. To address this problem the performance of QLCP is tested in hybrid networks and configured with special attention given edge nodes.

5.2 Motivation For Simulations

In the homogeneous configured network layout schemes, the RWS-AGE SBC mechanism gave the best performance in terms of least packet loss, low end-to-end delay, few collisions and improved fairness. The use of RWS-AGE is investigated in this section in hybrid configured network layout. The QLCP congestion control mechanism performed better over CCFS and is also investigated in this section in hybrid configured network layout. The mechanism of QLCP is specifically designed for edge nodes to control congestion. The hybrid configured network layout with different scheduling strategies are used to investigate the following research questions based on the hypotheses:

- Question 1: Does RWS-AGE perform better in homogeneous configured network layouts or hybrid configured network layouts in terms of packet loss and end-to-end delay?
- Question 2: Does QLCP perform better in homogeneous configured network layouts or hybrid configured network layouts in terms of packet loss and end-to-end delay?
- Question 3: Does the use of DCF in hybrid configured network layouts improve performance?
- Question 4: Which overall design combination gives the best performance in multi-domain networks?

5.3 Simulation Setup

The topology used for the testing of the hybrid configured network layouts is shown in figure 5.2. In this topology a clear distinction can be made between edge and core nodes in a network layout and is therefore, used for testing the hybrid configured network layout.

The hybrid performance investigations are carried out with EDCA, DCF, RWS-AGE and QLCP. The hybrid test layouts combinations are shown in table 5.1. QLCP is designed for edge nodes and therefore, it's performance in edge nodes is only investigated. The homogeneous configured network layout implementation performance results of EDCA, DCF, RWS-AGE and QLCP were presented in section 4.8.1 in chapter 4.

The same simulation environment and settings were used as those used for the homogeneous configured network layouts presented in section 4.3. Also the same performance metrics of end-to-end delay, packet loss, JFI and number of collisions are used to assess the performance of these hybrid configured network layouts.

5.4 Results and Discussion

This section presents the results of the scheduling strategies in the hybrid configured network layout. Section 4.8.1 presents the results of the strategies in the homogeneous configured network

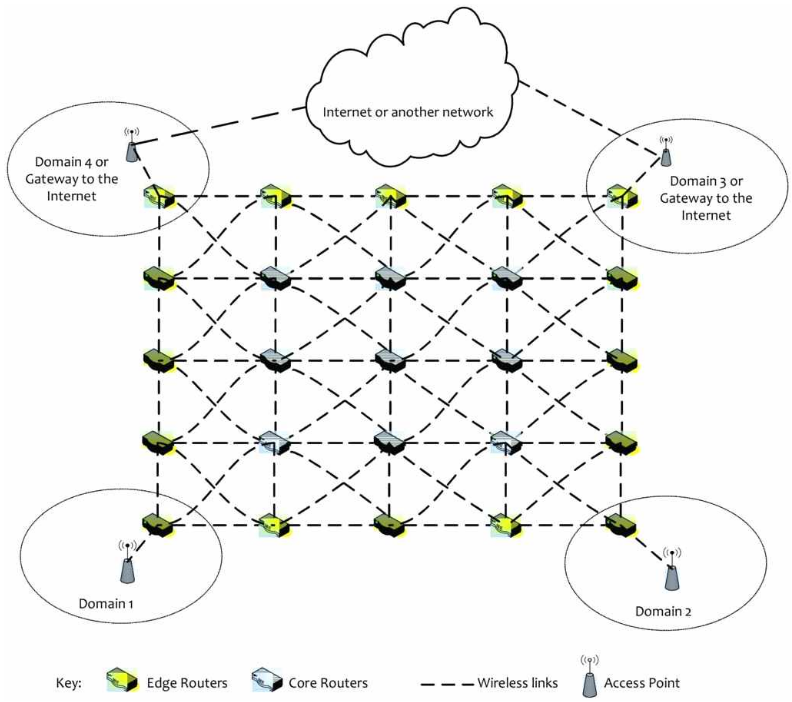


Figure 5.2: Multi-domain network which connect to a 5 x 5 wireless backbone mesh network.

Table 5.1: Hybrid configured network layout investigated.

	Edge Nodes	**Core Nodes**
Layout 1	DCF	EDCA
Layout 2	EDCA	DCF
Layout 3	RWS-AGE	DCF
Layout 4	QLCP	DCF
Layout 5	QLCP	RWS-AGE
Layout 6	EDCA	RWS-AGE
Layout 7	DCF	RWS-AGE

layouts. The homogeneous configured network layouts design layout results of EDCA, DCF, RWS-AGE and QLCP are also presented in this section for comparison.

5.4.1 Collisions

Figure 5.3 presents the graphs showing the total number of collisions in the network per *ms* with the different hybrid configured network layout. Hybrid configured network layout with DCF in the core nodes give all packets in the backbone network an equal chance to access the channel and are treated in a FIFO (first in first out) fashion. Using DCF in the hybrid configured network layouts reduces the collision probability as DCF uses a larger CW range for all the priority traffic types. Having a large CW range to choose a number for the back-off duration, reduces the changes of two nodes selecting the same back-off interval. The hybrid designs that use DCF are seen to experience the lowest collisions. Layout 5 (QLCP(E), RWSA(C)) and layout 6 (EDCA(E), RWSA(C)) are seen to experience the most collisions. This shows that DCF helps to lower collisions in a network.

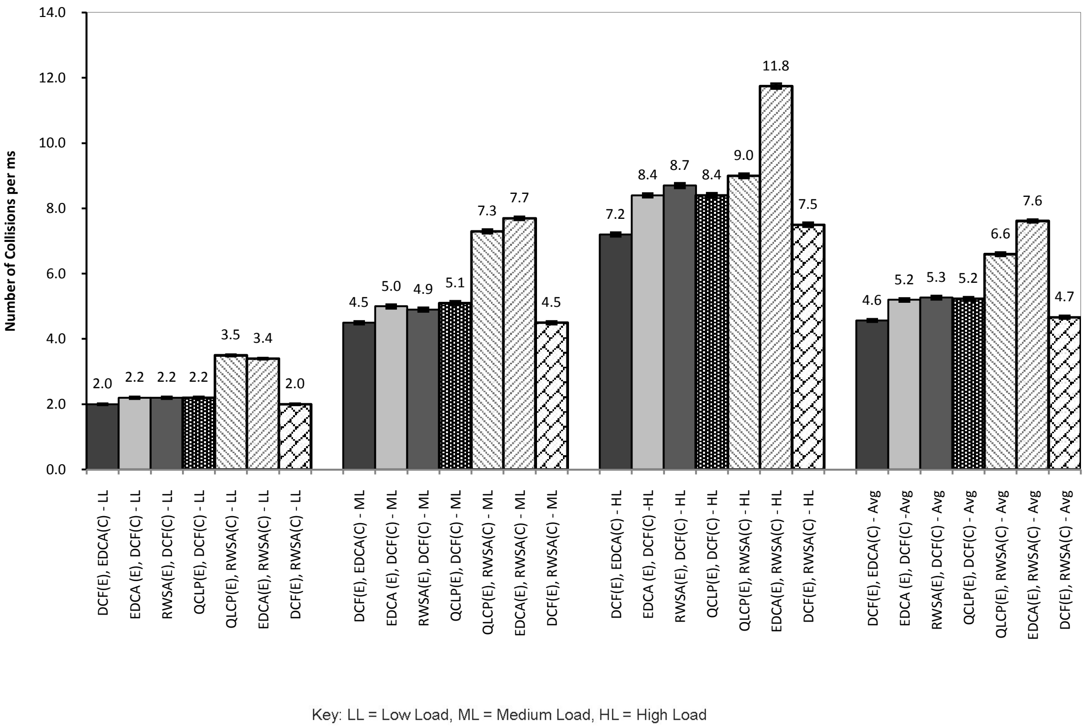


Figure 5.3: Average Number of Collisions with the different hybrid configured network layout scheduling strategies.

5.4.2 Packet Loss

Table 5.2 presents the summary and average calculated packet loss for all the data priority classes in the hybrid configured network layout investigated. Table 5.3 presents the summary and average calculated packet loss for all the data priority classes in the homogeneous configured network layout. Figure 5.4 presents the graph showing the packet loss for each priority class data in the hybrid configured network layouts investigated and figure 5.5 presents the graph showing the average packet loss over all the data priority classes.

The notation (E) is used to refer to the edge nodes and (C) to the core nodes in the figure. RWS-AGE is also written as RWSA in short form notation in this section. The homogeneous configured network layout DCF layout experiences less packet loss than the homogeneous configured network layout EDCA layout in WMNs. Compared to EDCA, the lower priority data have a higher overall chance of transmission with RWS-AGE and QLCP which lowers the collision probability on the network. The lowering of the collision probability occurs due to the fact that lower priority data use wider CW ranges. RWS-AGE is shown to experience the lowest packet loss in high load homogeneous configured network layouts. For the hybrid test layouts, DCF has been shown to reduce packet loss in the core and edge nodes. QLCP has been shown to perform better than homogeneous configured network layouts in a hybrid networks with DCF (C) and QLCP (E) when considering packet loss.

Thus, this provides an answer to the research question whether QLCP performs better in homogeneous configured network layouts or hybrid configured network layouts. QLCP experiences 29.8% packet loss on average data in the DCF (C) and QLCP (E) settings compared to 41.6% packet loss on average in its homogeneous configured network layout implementation under high loads. QLCP performs better in the hybrid setting with QLCP configured in the edge nodes and DCF in the core nodes with a packet loss reduction of 11.8% on average.

Considering the hybrid design investigated, having RWS-AGE in the core nodes, the layout with QLCP in the edge nodes showed high packet loss, while with DCF in the edge nodes showed the least packet loss. With DCF in the edge nodes, the packets are fed to the backbone network in a FIFO manner which also results in overall different types of packets in the backbone network having different CW ranges, thus reducing collisions. The design layouts with DCF in the edge and RWS or EDCA in the core nodes showed the least packet loss for HP data. This is not surprising as it is expected that a higher number of packets will successfully reach the core networks when DCF is used in the edge nodes, compared to using other priority selection queue strategies with shorter CW ranges. Since the average packet loss of these two cases layout 1 (DCF (E), EDCA (C)) and layout 7 (DCF(E), RWS-A(C)) are close, we apply the paired T-Test making a null hypothesis that the means are the same and an alternate hypothesis that the means are different. For packet loss the calculated T value is 2.00513 which is less than the T critical value of 2.144. Were therefore, accept the null hypothesis. The difference is not significant at a 95% confidence interval. RWS-AGE has been shown to experience the least packet loss on average (27.8%) in the hybrid layout 7(DCF(E) and RWS-AGE(C)), compared to its homogeneous configured network layout performance having a packet loss of 36.4% on average under high load. However, on average, over all the priority data classes, RWS-AGE in the homogeneous configured network layout experiences the least packet loss of 36.4% on average under high load.

Thus, this provides the answer to the research question whether RWS-AGE performs better in homogeneous configured network layouts or hybrid homogeneous configured network layouts. RWS-AGE thus performs well in both hybrid and homogeneous configured network layouts. However, the performance in better in its hybrid design implementations with layout 7(DCF(C), RWS-AGE(E)) with an average packet loss of 27.8% under high load or layout 3 (DCF(E), RWS-

Table 5.2: Summary of packet Loss with the different hybrid configured network layout scheduling strategies.

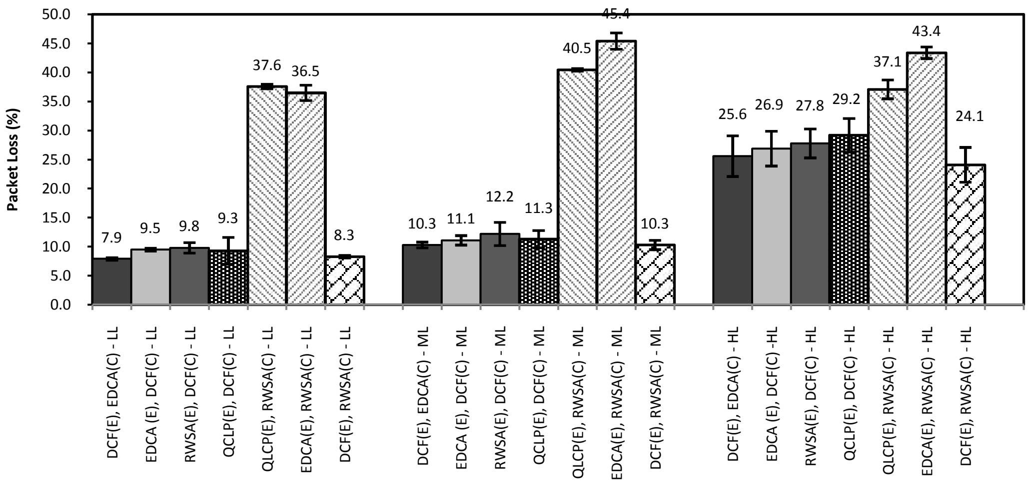
		HP (%)	MP (%)	LP (%)	Average (%)
Low Load	DCF(E), EDCA(C)	7.9	7.1	6.3	7.1
	EDCA (E), DCF(C)	9.5	11.0	5.9	8.8
	RWSA(E), DCF(C)	9.8	11.5	5.8	9.0
	QCLP(E), DCF(C)	9.3	11.2	5.8	8.8
	QLCP(E), RWSA(C)	37.6	24.2	5.8	22.5
	EDCA(E), RWSA(C)	36.5	26.8	5.8	23.0
	DCF(E), RWSA(C)	8.3	7.2	6.4	7.3
Medium Load	DCF(E), EDCA(C)	10.3	10.0	9.6	10.0
	EDCA (E), DCF(C)	11.1	11.8	8.4	10.4
	RWSA(E), DCF(C)	12.2	13.5	9.7	11.8
	QCLP(E), DCF(C)	11.3	11.6	7.7	10.2
	QLCP(E), RWSA(C)	40.5	27.8	10.2	26.2
	EDCA(E), RWSA(C)	45.4	35.1	14.4	31.6
	DCF(E), RWSA(C)	10.3	9.8	9.5	9.9
High Load	DCF(E), EDCA(C)	25.6	29.6	32.0	29.1
	EDCA (E), DCF(C)	26.9	27.7	30.9	28.5
	RWSA(E), DCF(C)	27.8	28.5	29.5	28.6
	QCLP(E), DCF(C)	29.2	29.7	30.6	29.8
	QLCP(E), RWSA(C)	37.1	36.4	29.7	34.4
	EDCA(E), RWSA(C)	43.4	38.2	45.7	42.4
	DCF(E), RWSA(C)	24.1	28.4	31.0	27.8

Table 5.3: Summary of packet Loss with the different homogeneous configured network layout scheduling strategies under study.

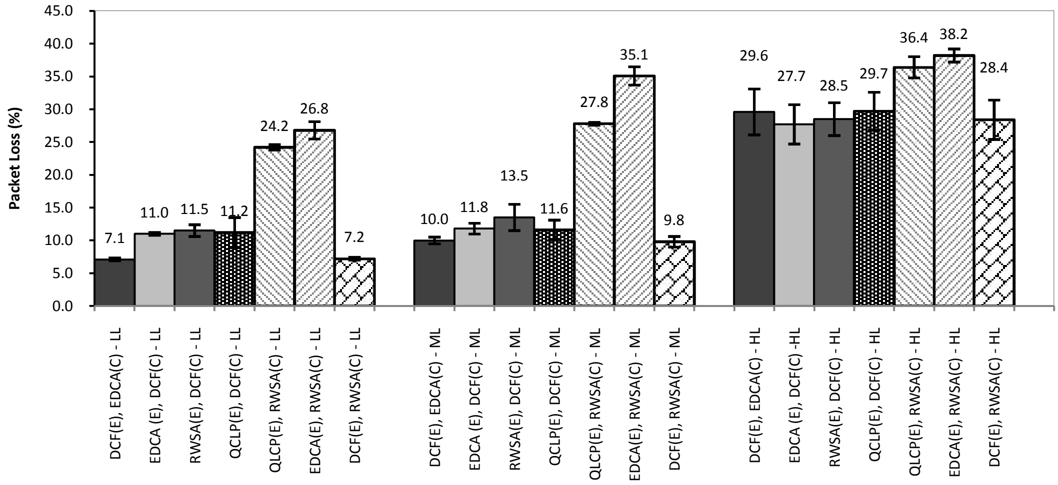
		HP (%)	MP (%)	LP (%)	Average (%)
Low Load	EDCA	41.1	29.9	5.9	25.6
	DCF	6.3	6.3	6.3	6.3
	RWS-AGE	24.9	23.8	6.0	18.2
	QLCP	41.1	25.7	6.6	24.5
Medium Load	EDCA	45.2	35	14.7	31.6
	DCF	15.3	15.8	15.5	15.5
	RWS-AGE	17.4	17.5	11.7	15.5
	QLCP	39.4	27.2	10.4	25.7
High Load	EDCA	46.4	39.8	57.9	48.0
	DCF	48.4	48.4	48.8	48.5
	RWS-AGE	35.3	38.7	35.1	36.4
	QLCP	41.7	42.6	40.5	41.6

AGE(C))with an average packet loss of 28.5% under high load.

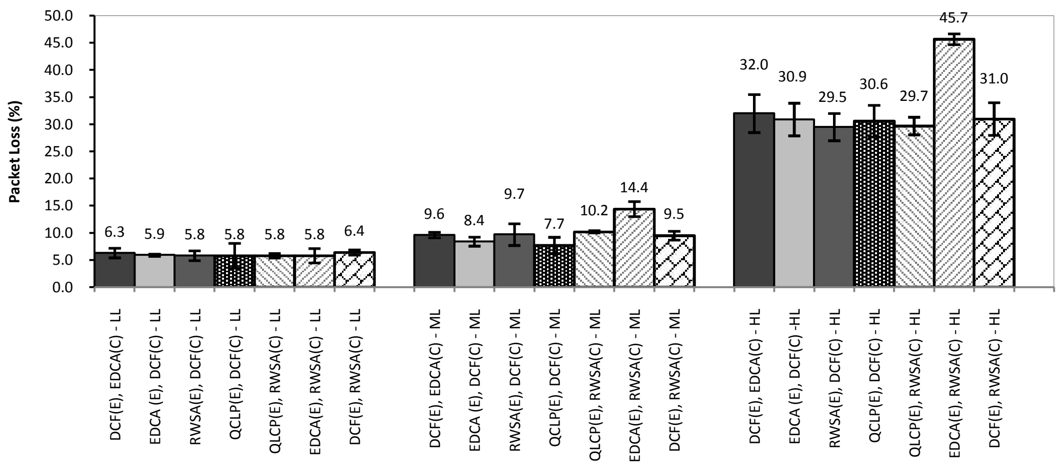
The use of DCF in hybrid layouts has also shown on average to lower packet loss. This provides also our answer to the research question whether the use of DCF in hybrid networks improve performance. The hybrid design layouts DCF(E) EDCA(C), EDCA(E)DCF(C), RWSA(E) DCF(C), QLCP(E) DCF(C) and DCF(E) RWSA(C) have all experienced lower packet loss than their homogeneous configured network layout implementations. The use of DCF in hybrid layouts has been shown to be an important ingredient to lowering packet loss.



(a) High Priority Data



(b) Medium Priority Data



(c) Low Priority Data

Figure 5.4: Packet loss with the different hybrid design layout scheduling strategies.

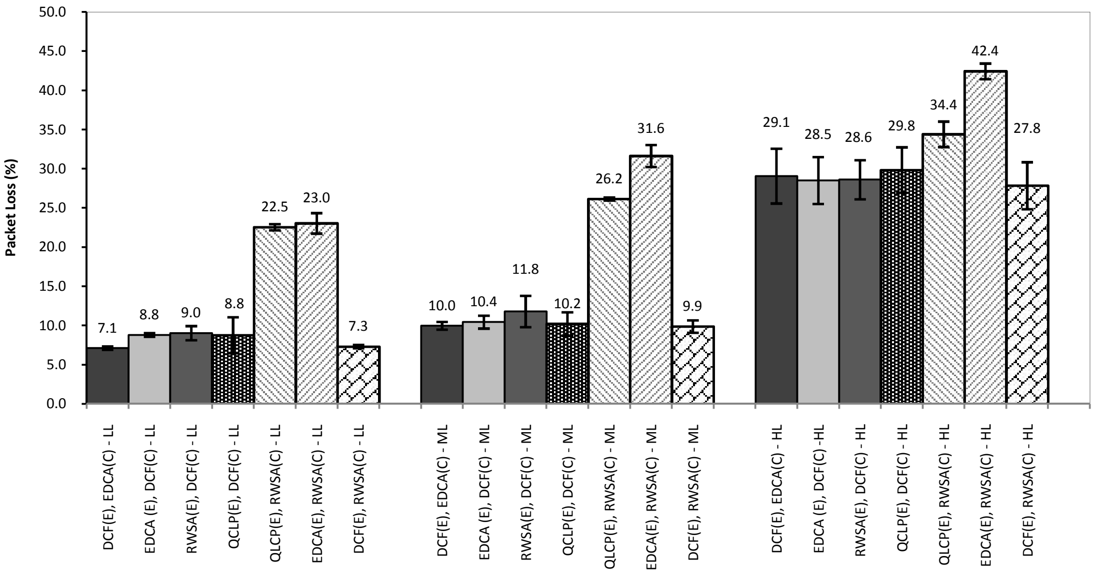


Figure 5.5: Average Packet loss with the different hybrid configured network layout scheduling strategies.

5.4.3 End-to-End Delay

Figure 5.6 presents the graph showing the end-to-end delay experienced for each priority class data in the hybrid configured network layouts investigated and figure 5.7 presents the graph showing the average calculated end-to-end delay experienced over all the data priority classes. Table 5.4 presents the summary of the average calculated end-to-end delay as well as for the individual priority class end-to-end delay for the hybrid design layouts. Table 5.5 presents the average calculated end-to-end delay as well as for the individual priority class end-to-end delay for the homogeneous configured network layouts. The lowest end-to-end delay is experienced with the use of the homogeneous configured network EDCA layout in medium and high loads as the higher priority data packets have the smallest contention periods. RWS-AGE in the homogeneous configured network layout experiences more end-to-end delay than the homogeneous configured network EDCA layout for HP and MP data, but less than the homogeneous configured network DCF layout on average as with RWS-AGE, as a higher number of other priority packets will be transmitting on the network compared to the homogeneous configured network EDCA layout, thus increasing the end-to-end delay. QLCP in the homogeneous configured network layout also experiences less end-to-end delay than the homogeneous configured network DCF layout, but more than the homogeneous configured network EDCA layout for the same reasons as with RWS-AGE.

In section 5.4.2, a conclusion was drawn based on the paired T-Test that there the packet loss mean difference is not significant at a 95% confidence interval between layout 1 (DCF(E), EDCA(C)) and layout 7 (DCF(E),RWS-A(C)). We apply the paired T-Test on the end-to-end delay scenarios. For end-to-end delay the calculated T value is 2 for these data rates which is again also less than the T critical value of 2.144. We therefore, accept the null hypothesis. The difference is not significant at a 95% confidence interval for end-to-end delay. However, based on fact that the mean packet loss for HP data and on average is lower in layout 7 than 1, layout 7 (DCF(E), RWS-A(C) is chosen over layout 1(DCF (E), EDCA (C)) as a better choice. This is also based on the reasoning that in the homogeneous configured network layout, the RWS-AGE strategy does not starve lower priority data as compared to EDCA and also experiences less packet loss.

Results show that the hybrid schemes where QLCP is configured in the edge nodes that are subjected to more load compared to other nodes in the network help reduce end-to-end delay. With the hybrid schemes, based on the results of HP data and the average end-to-end delay, it can be seen that design layout 4 (QLCP (E), DCF (C)) and layout 2 (EDCA(E), DCF(C)) showed both less packet loss and low end-to-end delay. To determine if there is a statistical difference in their means, we carry out the paired T-Test making a null hypothesis that the means are the same and an alternate hypothesis that the means are different. For packet loss the calculated T value is 2.62 which is greater than the T critical value of 2.144. Were therefore, reject the null hypothesis. For end-to-end delay the calculated T value is 3.52 which is greater than the T critical value of 2.144. Were therefore, reject the null hypothesis again. Therefore, layout 2 (EDCA(E), DCF(C)) performs better than layout 4 (QLCP (E), DCF (C)) under high loads. Under continuous heavy load scenarios for HP data, QLCP starves lower priority data if it does not have a queue length more than the threshold, while with EDCA, the lower priority will continuously be counting down and keep trying to access the medium. In such a scenario, there is a higher chance with EDCA than QLCP for the lower priority data to transmit. The results show that QLCP performs better in terms of end-to-end delay in a hybrid setting of QLCP(E) and DCF (C) with a reduction of 33.9 ms on average compared to its homogeneous configured network layout implementation.

Thus, this answers our research question on whether QLCP performs better in homogeneous

Table 5.4: Summary of End-to-End Delay with the different hybrid configured network layout scheduling strategies.

		HP (ms)	**MP** (ms)	**LP** (ms)	**Average** (ms)
Low Load	DCF(E), EDCA(C)	4.8	7.3	4.5	5.5
	EDCA (E), DCF(C)	4.1	5.3	4.0	4.5
	RWSA(E), DCF(C)	4.8	4.9	4.0	4.6
	QCLP(E), DCF(C)	3.8	5.6	4.0	4.5
	QLCP(E), RWSA(C)	5.6	7.9	5.9	6.5
	EDCA(E), RWSA(C)	6.0	8.0	5.9	6.6
	DCF(E), RWSA(C)	4.8	7.2	4.5	5.5
Medium Load	DCF(E), EDCA(C)	5.4	8.3	7.4	7.0
	EDCA (E), DCF(C)	4.9	6.2	6.3	5.8
	RWSA(E), DCF(C)	7.2	7.1	6.4	6.9
	QCLP(E), DCF(C)	4.7	6.6	6.7	6.0
	QLCP(E), RWSA(C)	6.1	9.2	16.5	10.6
	EDCA(E), RWSA(C)	5.4	8.2	16.1	9.9
	DCF(E), RWSA(C)	5.4	8.3	7.5	7.1
High Load	DCF(E), EDCA(C)	21.0	24.4	31.1	25.5
	EDCA (E), DCF(C)	16.6	18.5	22.6	19.2
	RWSA(E), DCF(C)	24.6	24.0	23.4	24.0
	QCLP(E), DCF(C)	17.5	20.4	24.9	20.9
	QLCP(E), RWSA(C)	24.4	28.9	74.4	42.6
	EDCA(E), RWSA(C)	5.4	8.9	89.3	34.5
	DCF(E), RWSA(C)	21.8	25.2	32.3	26.4

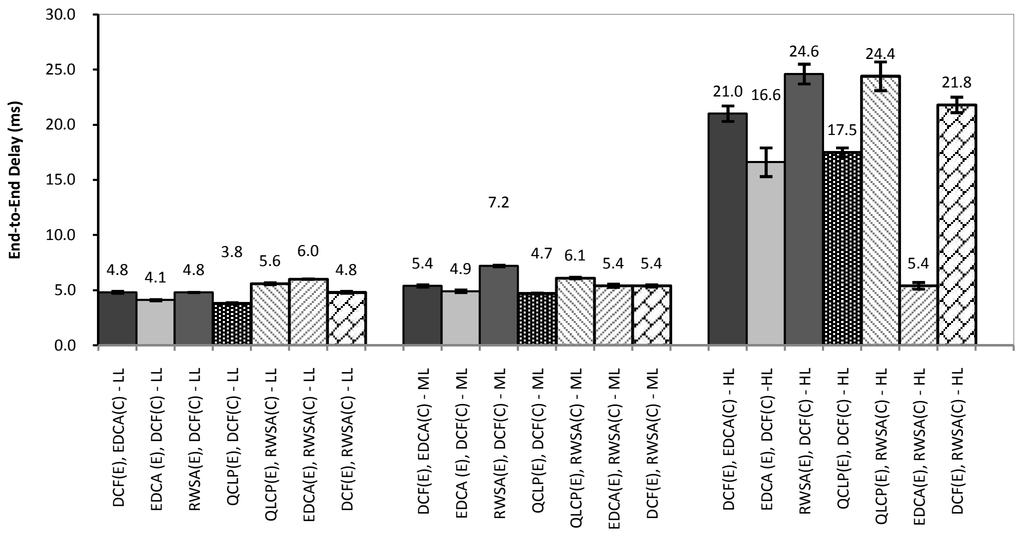
or hybrid settings. QLCP performs better in terms of lower end-to-end delay in the QLCP(E), DCF (C) layout. This supports the hypothesis that a load control scheduling strategy for gateway nodes that are subjected to more traffic load can reduce packet loss in a hybrid design layout where different nodes are assigned different scheduling strategies with some of these devices assigned DCF.

The results also show that RWS-AGE performs better in terms of end-to-end delay in a hybrid setting of layout layout 7 (DCF(E) and RWS-AGE (C)) or layout 3 (RWS-AGE(E), DCF(C)) compared to its homogeneous configured network layout implementation for HP data only. Layout 3 experiences less end-to-end delay than layout 7.

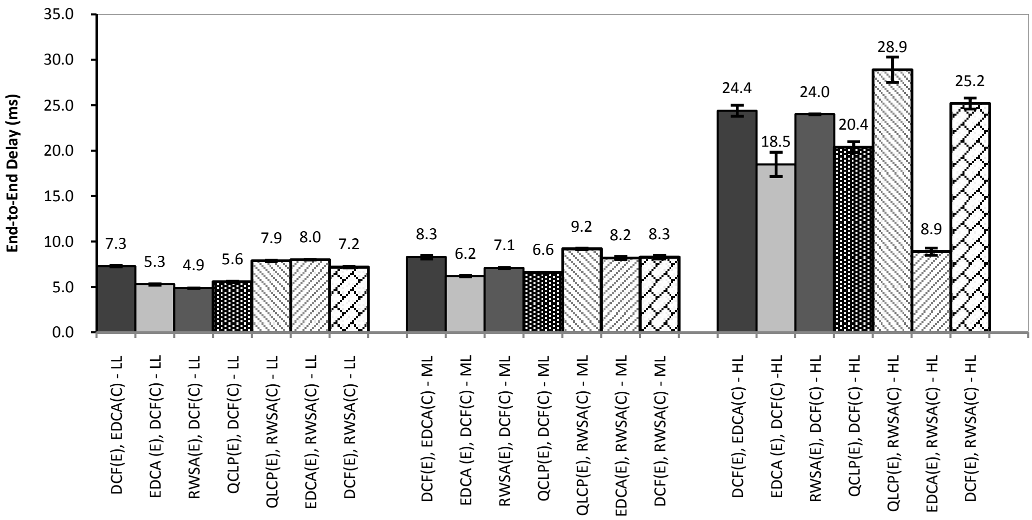
Between layout 1 (DCF(E), EDCA(C)) and layout 2(EDCA(E), DCF(C)), layout 2 experiences lower end-to-end delay on average. It can be concluded that with the use of DCF in the core, less end-to-end delay is experienced compared to when it is used in the edge nodes.

Table 5.5: Summary of End-to-End Delay with the different homogeneous configured network layout scheduling strategies.

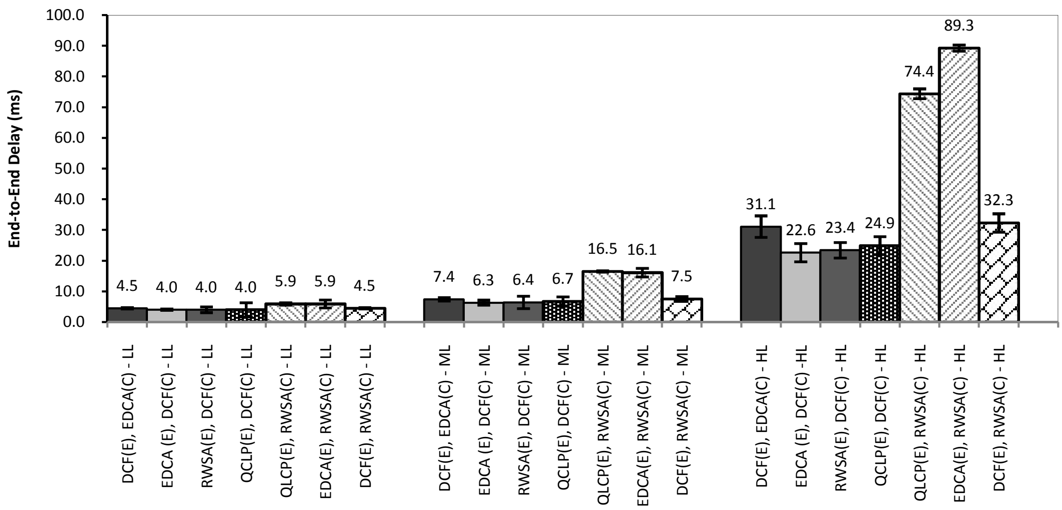
		HP (ms)	**MP** (ms)	**LP** (ms)	**Average** (ms)
Low Load	EDCA	5.3	7.8	5.9	6.3
	DCF	11.0	11.8	11.7	11.5
	RWS-AGE	6.1	7.4	5.8	6.4
	QLCP	5.1	8.1	5.9	6.4
Medium Load	EDCA	5.3	8.1	16.8	10.1
	DCF	37.3	37.3	37.1	37.2
	RWS-AGE	10.5	15.0	16.2	13.9
	QLCP	6.4	9.6	19	11.7
High Load	EDCA	5.6	9.1	143.2	52.6
	DCF	61.4	61.2	61.4	61.3
	RWS-AGE	28.1	62.9	58.7	49.9
	QLCP	30.9	35.8	97.7	54.8



(a) High Priority Data



(b) Medium Priority Data



(c) Low Priority Data

Figure 5.6: End-to-End Delay with the different hybrid design layout scheduling strategies.

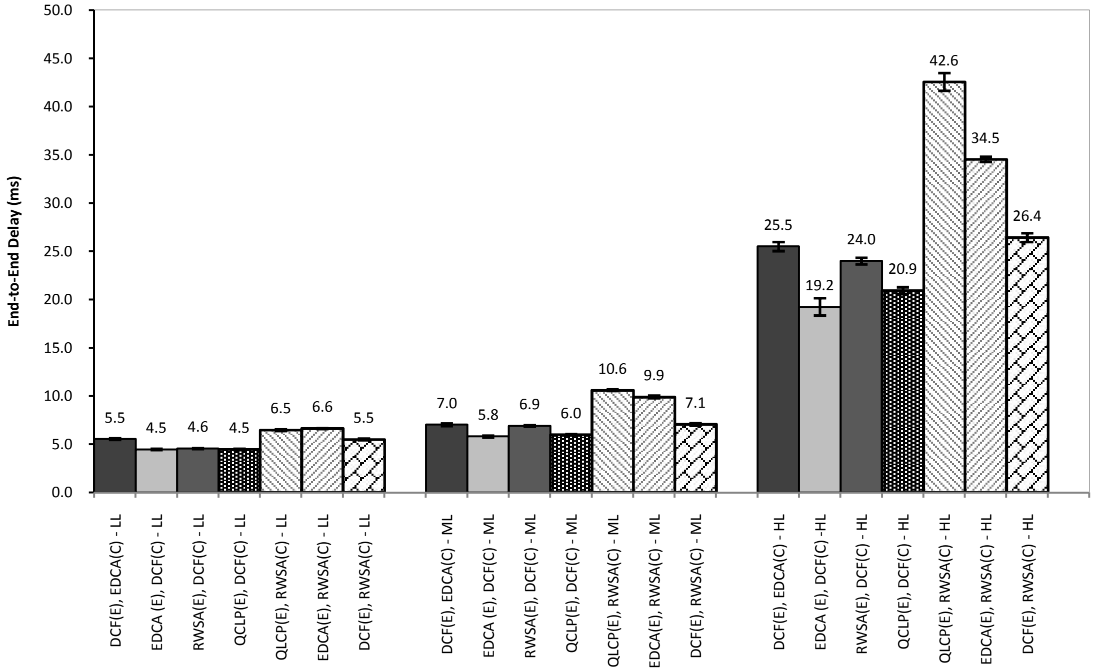


Figure 5.7: Average End-to-End Delay with the different hybrid configured network layout scheduling strategies.

5.4.4 Jain's Fairness Index

Figures 5.8 present the Jain's fairness index for the heavy load scenarios for the hybrid configured network layouts. All the hybrid design layouts investigated tend to improve fairness under heavy loads. Layout 2 and layout 4 provide the highest fairness under heavy loads. When DCF is used in the core nodes, a higher improvement in fairness is observed compared to when DCF is used in the edge nodes.

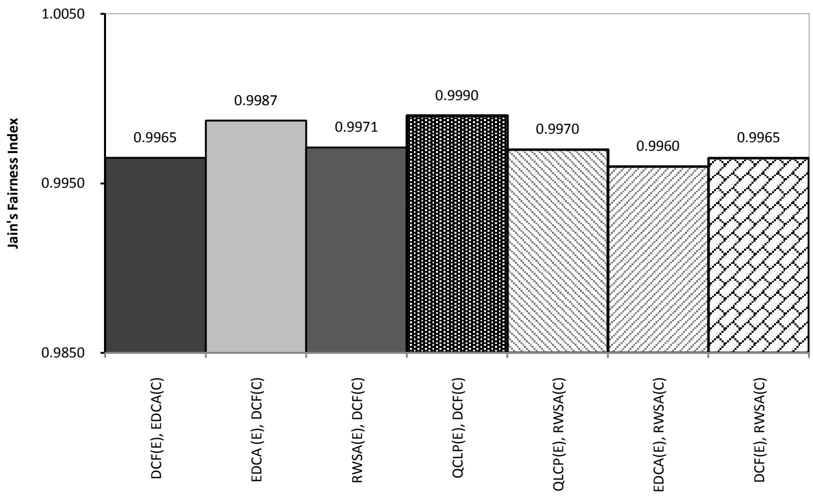


Figure 5.8: Fairness under heavy load with the different hybrid configured network layout scheduling strategies

5.4.5 Conclusion

The choice of the scheduling strategy will depend largely on the architectural layout and functioning of the network. Homogeneous configured network DCF design layouts provide a higher degree of reliability (less packet loss) compared to homogeneous configured network EDCA layouts. However, DCF results in high end-to-end delay compared to EDCA for higher priority data. With homogeneous configured network RWS-AGE layouts, the overall reliability is improved compared to the homogeneous configured network EDCA and DCF layouts. However, the end-to-end delay is less than the homogeneous configured network DCF layout, but more than the homogeneous configured network EDCA layout. With QLCP, the performance is not optimum in homogeneous configured network layouts.

Hybrid design layouts with DCF presented improved performance over their homogeneous configured network layout implementations. The different priority traffic carried in the backbone nodes configured with DCF gain access to the medium in a FIFO manner with same channel access parameter values. The edge nodes configured with EDCA or QLCP give a higher chance to the higher priority data to access the network. The layout where EDCA was configured in the edge nodes and DCF in the core nodes experienced the best performance in terms on less packet loss and least end-to-end delay. In these hybrid design schemes, the core nodes are basically performing non-differentiated data services on a FIFO basis and the edge nodes are performing data differentiated services according to the data priority. RWS-AGE is not designed for congestion control and performs well in both homogeneous configured network layouts and hybrid configured network layouts. The hybrid layout with DCF configured in the edge nodes and RWS-AGE configured in the core nodes showed the least packet loss and low end-to-end delay while the hybrid layout with EDCA configured in the edge nodes and DCF configured in the core nodes showed the least end-to-end delay and low packet loss.

This provides the answer to our research question on which hybrid configured network layout performs the best. The hybrid configured network layout with DCF (E), RWS-AGE (C) showed the least packet loss and low end-to-end delay while the hybrid configured network layout with EDCA (E), DCF (C) showed the least end-to-end delay and low packet loss. The hybrid strategy with DCF in the edge nodes and RWS-AGE in the core nodes out performs the other investigated homogeneous and hybrid strategies in terms of packet loss. The use of DCF in the edge nodes only increases the end-to-end delay compared to when it is used in the core nodes only.

The network layout and design plays a critical role in determining the choice of scheduling strategies. Networks that require high reliability, but can tolerate slightly more end-to-end delay, a hybrid configured network layout, where DCF is configured in the edge nodes and EDCA is configured in edge nodes will be a good design to use if the network has gateway nodes. Networks that require low end-to-end delay and can tolerate slight packet loss, a hybrid configured network layout, where DCF is configured in the core nodes and EDCA is configured in edge nodes will be a good design choice. The results have shown that the choice of the scheduling strategy taking the network design into account is an important ingredient in coordinating access to the medium in an effective manner to achieving efficient QoS in SRSC WMNs.

Chapter 6

Testbed Implementation

6.1 Introduction

The FIT IoT-lab testbed at Inria was used to implement and test the RWS-AGE scheduling strategy. The testbed testing was done to ascertain that there is a packet loss reduction over the DCF strategy. It has been shown in chapter 4 that with DCF in multi-hop networks, less packet loss is experienced when compared to EDCA. Wireless sensor networks (WSN) form part of WMNs. WSN embedded operating systems include among others TinyOS, Contiki, MANTIS, T-Kernal, LiteOS and Nano-RK. The Contiki operating system is one of the dominant operating systems for embedded systems and for IoT applications [122]. Contiki works with the default CSMA/CA MAC scheduling strategy. This chapter presents an overview of the testbed implementations carried out and the results obtained.

6.2 Overview of CSMA/CA in IEEE 802.15.4 Standard

This section presents an overview of the CSMA/CA in the Contiki operating system. One of the main driving forces of the features of machine-to-machine communication and Internet of things (IoT) has been the wide area of research leading to development of low-power, low-rate, and low-cost wireless systems. The IEEE 802.15.4 standard has become a standard for these low rate wireless personal area network (LR-WPAN) communications [123]. The IEEE 802.15.4 standard, which operates at the link-layer and physical layer, is designed for simple, low data rate, low-power and low-cost wireless communication with wireless personal area networks (WPANs). This standard is mainly implemented in wireless sensor networks. The unlicensed ISM band that operates worldwide with this technology is the 2.4 GHz ISM band [124]. In this band of 2.4 GHz, the ISM offers 16 channels with a data rate of 250 kbps [125]. Wireless data exchange is done through the DSSS modulation scheme [124]. In the implementation, nodes with a radio technology that use the 2.4 GHz ISM band are used.

According to the IEEE 802.15.4 standard protocol, a node can optionally operate in beacon-enabled mode or non-beacon enabled mode [125]. In this section, we present a brief overview of the non-beacon enabled mode CSMA/CA mechanism of the IEEE 802.15.4 standard which is based on the un-slotted mode used in the implementation. The slotted mode requires slot synchronization.

When a packet arrives, the number of backoffs (NB) and the backoff exponent (BE) are initialized. BE is the backoff exponent which is the number of back-off periods that a device should wait before attempting to assess the channel. NB is the number of backoffs. After this initialization of the variables, the back-off period is started which is chosen by a random number generated in the range of $[0, 2^{BE}\text{-}1]$. Initially, BE is initialized to BE_{min} which is 3 by default. BE_{max} is 5 by default. When this back-off has expired, the algorithm then performs one Clear Channel Assessment (CCA) to verify if the channel is busy or free. If the channel is found to be busy, the NB and BE variables are incremented by one. The procedure is repeated until NB is less than the set maximum allowed transmissions. If the channel is found to be free (idle), a transmission takes place, otherwise the packet is dropped.

In the IEEE 802.15.4 standard, the acknowledgement (ACK) mode is optional unlike in the IEEE 802.11 standard. It is an optional feature as it can increase network overhead and have an effect on the achievable throughput of the network. If ACK mode is enabled, for any transmission that does not receive an acknowledgment, the NB and BE values are increased. If NB becomes greater than the maximum transmissions allowed, the algorithm terminates with a channel access failure status [126]. Figure 6.1 presents the flowchart for the operation of CSMA/CA in non-beacon unslotted mode.

6.3 Contiki

This section presents a brief overview of this Contiki operating system which is an open source operating system. Contiki is implemented in the C language developed at the Swedish Institute of Computer Science (SICS) [127,128]. Contiki has an event-driven kernel and follows a linear programming style which was also used for the programming in this work. The Contiki protocol stack is designed for resource-constrained devices with constraints on memory and processing power [129]. It supports IPv6; RPL routing protocol for low-power and lossy networks; Rime and the Constrained Application Protocol (CoAP), making it suitable to develop IoT applications [122]. Compared to many other closed source firmware operating systems implemented in hardware, Contiki is open source. We therefore, used Contiki in our testbed implementation as it allows us to use and modify existing codes.

The Contiki OS provides two communication stacks namely uIP and Rime. uIP is a TCP/IP stack that makes it possible for Contiki to communicate over the Internet. Rime is a lightweight communication stack designed for low-power radio communication. Rime is a custom lightweight networking stack with lower overhead compared to uIP. It provides primitives for single-hop and multi-hop (mesh) communication [128]. In the implementation, the Rime communication stack for multi-hop communication was used as the other layers are less detailed. Figure 6.2 presents the communication protocol stack used in our study. At the physical layer, the 2.4 GHz radio module is used. The standard ContikiMAC radio duty cycle is used. The RWS scheduling strategy is implemented at the MAC layer with CSMA/CA. The Rime routing is used at the network layer.

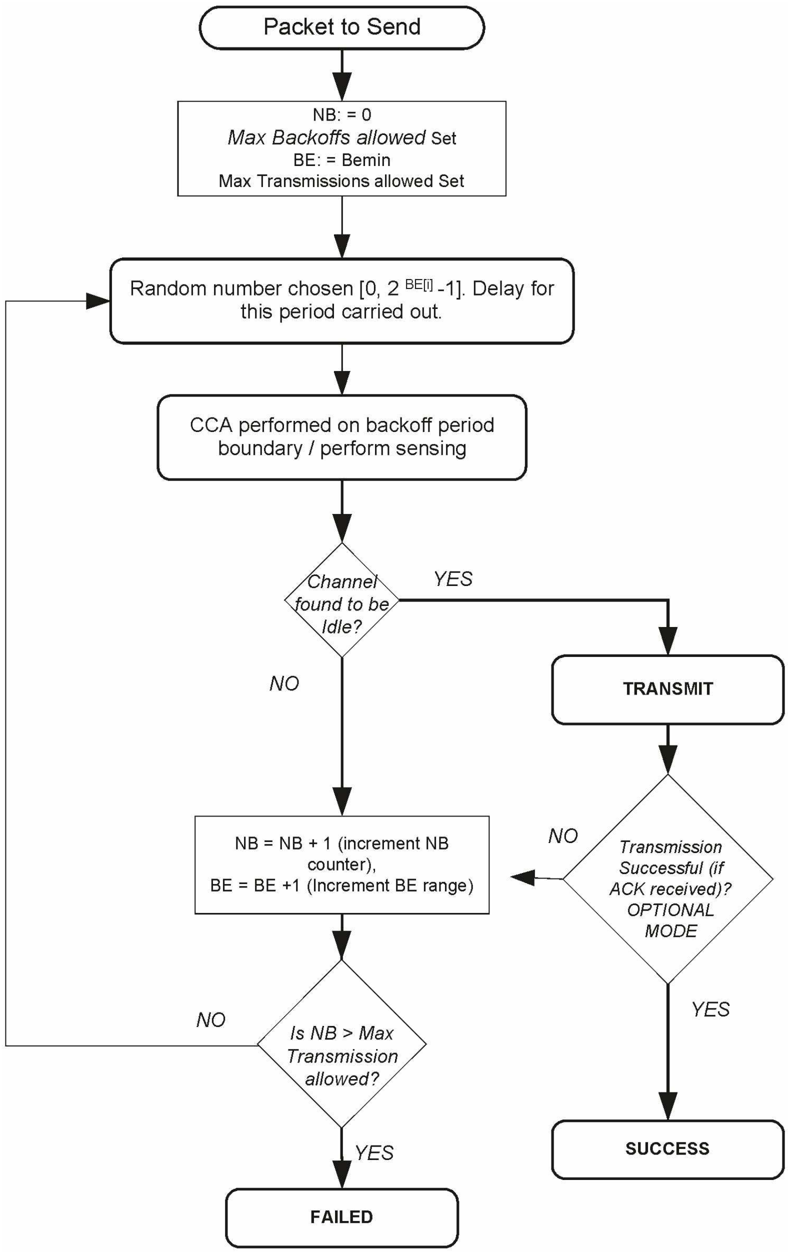


Figure 6.1: Flow chart for the non-beacon enabled un-slottted mode for the CSMA/CA mechanism with ACK.

6.4 Testbed Implementation

In the packets generated on the testbed, a data field is created of 2 bits which carries information on the priority of the packet. Using this information, data is placed in either one of the 3 queues depending on the priority set in the packet header. The priority field in the packet is shown in figure 6.3.

When a node has data to transmit and more than one queue has data, a selection scheduling strategy is followed. If only one queue has data, then the packet from that queue is selected for transmission without the need to follow a selection process. The BE and NB processes are carried out after the selection process. The BE values are assigned for the different priority queues to match the CW_{min}, CW_{max} and back-off process sizes used in chapter 4 as shown in table 6.2.

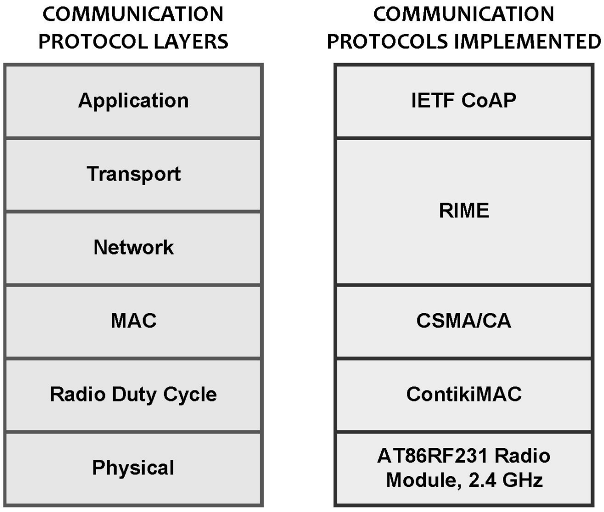


Figure 6.2: Contiki Layer Model.

Figure 6.3: Integration of priority information in the packet.

Table 6.1: RWS parameters

Traffic Type	**Minimum BE Value**	**Maximum BE Value**	**Minimum CW Value**	**Maximum CW Value**
High Priority Data	3	4	7	15
Medium Priority Data	4	5	15	31
Low Priority Data	5	10	31	1023

The implementation and testing of the scheduling strategy was done on nodes in Lille and Grenoble. The layout of the Lille M3 nodes is shown in figure 6.4 and for Grenoble in figure 6.5. The strategy was implemented on M3 nodes which use a 32-bit ARM cortex M3 micro-controller with 64 KB of RAM. They use a 2.4GHz radio interface.

The RWS scheduling strategy is mainly developed for backhaul nodes that will carry data of different priority in a multi-hop fashion until it reaches its destination. The Grenoble test bed site was used for testing of the line topology and Lille test bed site was used for testing of the grid topologies.

The mesh network was setup so that communication with the receiver takes place in multiple hop fashion by limiting the transmission range of the nodes. There are two ways of limiting the transmission range: (1) decreasing the transmission power and (2) by setting a minimal energy level for the packet reception. The range of the nodes was limited in the testbed by reducing the transmission power.

The default CSMA/CA scheduling strategy in Contiki works in a FIFO fashion and does not differentiate between packets of different priority. The RWS-AGE strategy was developed with CSMCA/CA by introducing three queues in the nodes, one for each priority level. The link

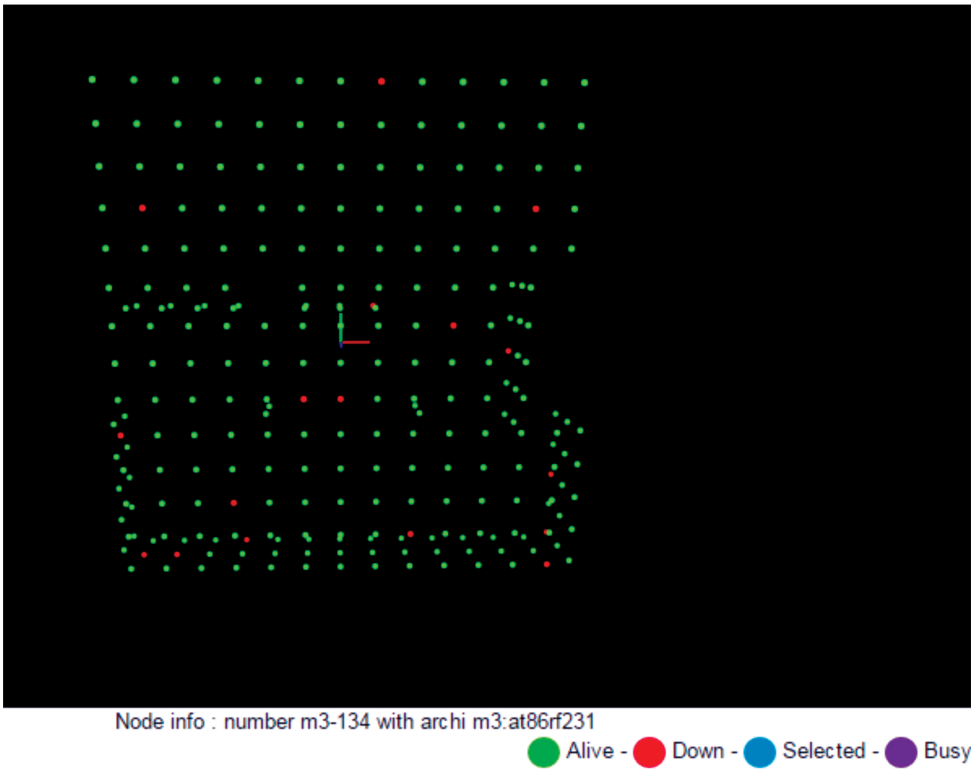


Figure 6.4: M3 nodes layout in Lille.

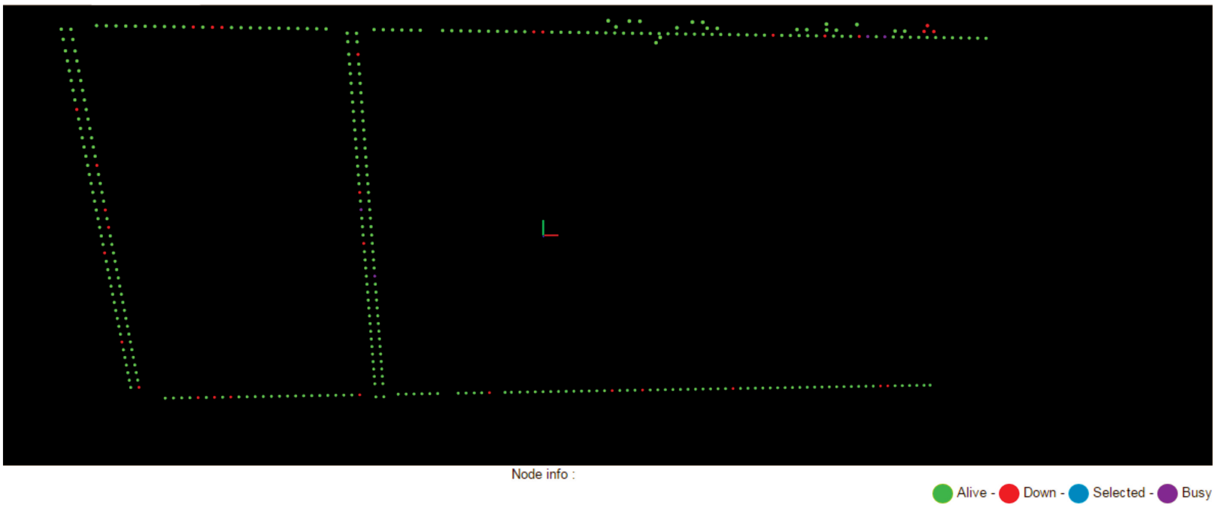


Figure 6.5: M3 nodes layout in Grenoble.

list library in Contiki manages packet queues. An application code was written that generates packets of 128 bytes of different priority levels at the transmission rates of the different test cases. Packets with the fields as shown in figure 6.3 were created. An application was written so that each node records the number of packets sent and number of packets received. Before implementation on the actual test-bed, the codes written in C were tested in the Cooja simulator on Tmote Sky nodes. After that, they were compiled for the actual testbed nodes and implemented on the testbed.

The parameters for BE and NB were adjusted to match those of the proposed scheduling

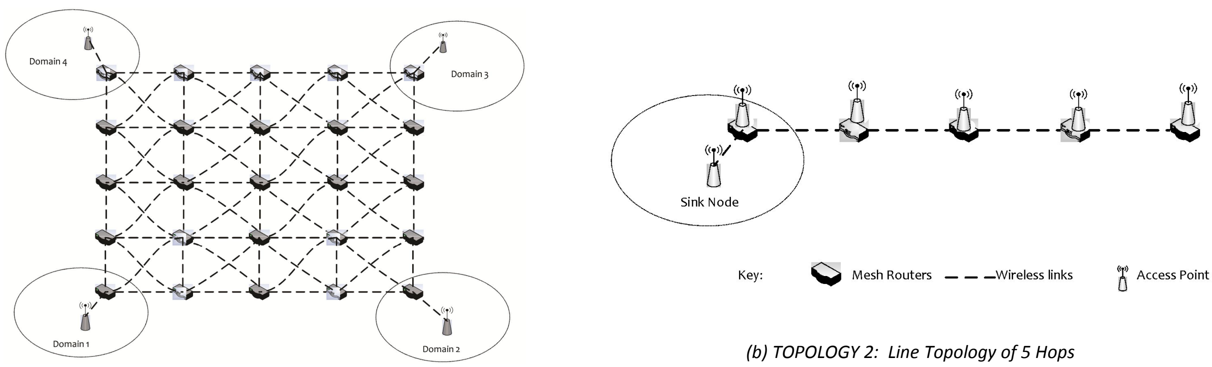
Table 6.2: Different load level test scenarios.

Load Level	**Normalised Offered Load**
Low	0.3
Medium	0.6
High	0.9

strategy in the IEEE 802.11 standard. The modifications therefore made to the CSMA/CA in the IEEE 802.15.4 were as follows:

- The acknowledgment mechanism was activated to receive acknowledgment messages for any successful transmission as in the IEEE 802.11 CSMA/CA.
- The maximum transmission value allowed was set to 7 as is the case with IEEE802.11g.
- The values of BE_{min} and BE_{max} were changed such that the CW size will be the same as in the EDCA based on the CW_{min} and CW_{max} values. For the RWS strategy, the values for each priority data category were changed according to the CW_{min} and CW_{max} values for the different categories as shown in Table 6.2.
- For DCF, BE_{min} is set to 5 as a BE value of 5 equals a CW size of 31 and BE_{max} is set to 10 as a BE value of 10 equals a CW size of 1023.

The test topologies are shown in figure 6.6. The proposed RWS strategy was tested against the original CSMA/CA. The load test cases are shown in table 6.3. The limiting of the transmission power for communication for 1 hop to be only possible (with the direct neighbours) was successfully done for topology 2 (line topology) for nodes in Grenoble. The nodes in Grenoble are spaced and outlying nodes could be easily used for this topology. However, in topology for implementation in Lille, the nodes chosen are close to each other to implement the topology. Although the transmission power was reduced as much as power to make only 1 hop range communication only possible, the chosen power still allowed 2 hop communication for a few packets.



(b) TOPOLOGY 2: Line Topology of 5 Hops

(a) TOPOLOGY 1: 5 by 5 Grid Topology

Figure 6.6: Testbed topology test cases.

The results in section 6.4 are presented in terms of packet loss. Since the measurements are made on a test bed, the end-to-end delay measurements could not be obtained accurately and are not presented. Two extra fields were added to the packets to obtain end-to-end delay.

One for recording the time the packet was sent and the other for when the packet was received. An application was written to subtract these values and obtain the average end-to-end delay over all the packets received. However, the clocks of the nodes are not completely synchronized on the test bed. To avoid this issue in the end-to-end delay results, the application was then written such that when the destination receives the packet from the source, it sends it back to the source. This will allow the measurements to be made from the same clock as a receiver sends the packet back to the source. This was the round trip time (RTT). Doing this resulted in the packet loss becoming extremely high as the contention on the network increased significantly. Also the end-to-end delay results were not accurate as the value would vary significantly when the experiment was repeated continuously.

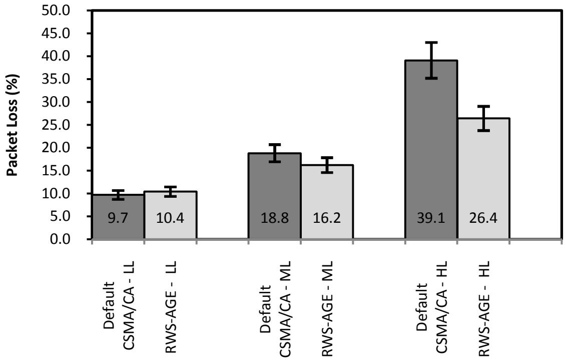
6.5 Results

The packet loss results for the different test cases with the default CSMA/CA with the proposed scheduling strategy are shown in figure 6.7 for the square grid topology and in figure 6.8 for line topology. The results are those obtained from the real test bed implementation as such the conditions of the channel can change over time depending on the environment. The performance of CSMA/CA depends also on the value chosen for the back-off which is randomly selected. For any two testbed tests carried out, the exact conditions might not be the same as the number generated might be different which effects when the packets are transmitted to the next hop as well as the link conditions. The proposed scheme also largely depends on a random number generated to choose which queue must transmit its data. This clearly shows that the results obtained for any two tests may not be under the same test conditions. To carry out the tests under the same link conditions of the channel, both tests for each hop number were run immediately one after the other for CSMA/CA and RWS-AGE for the same hop number to make the comparison approximately the same.

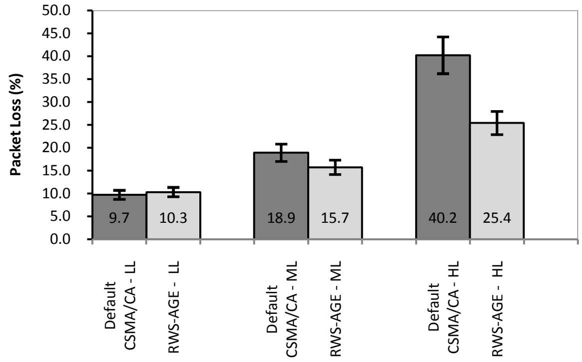
For all the three test topologies, the tests were run for five minutes each. An improvement in performance (less packet loss) can be observed for RWS-AGE over CSMA/CA under heavy loads. Higher packet loss is experienced for medium and heavy load test cases for topology 2 than topology 1. This could have also been due to the transmission power situation mentioned in section 6.4. EDCA was not implemented on the testbed over the IEEE 802.15.4 standard as the mechanism is more complex than this standard. It requires the back-off count down counter to freeze when a packet is being transmitted on the channel. The counter does not freeze in CSMA/CA in the IEEE 802.15.4 standard. It also requires an internal contention mechanism. If implemented, the strategy would have only been an approximate and not the exact strategy.

6.6 Conclusion

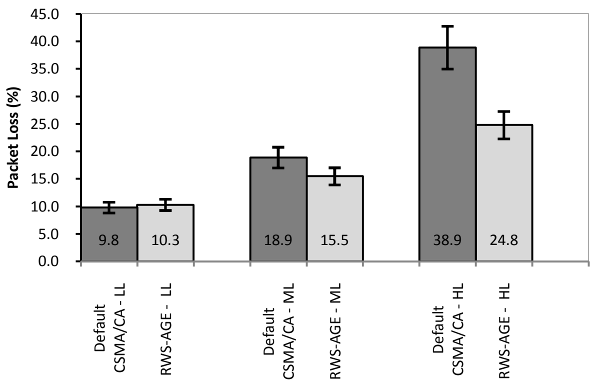
This chapter has presented an overview of the implementation carried out on the testbed. Contiki is an open source operating system and therefore, we modified the existing codes to implement our strategy. The Rime protocol communication stack was used as the other layers are light weight and this helps to ascertain the performance of the proposed scheme. The RWS-AGE scheduling strategy has shown a reduction in packet loss as the number of hops increase for the presented test cases implemented over the FIT IoT-lab testbed.



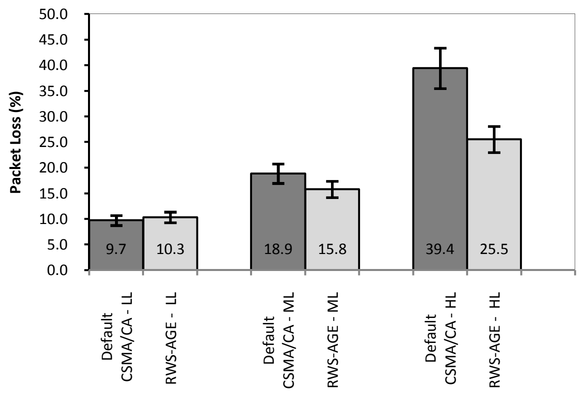
(a) High Priority



(b) Medium Priority

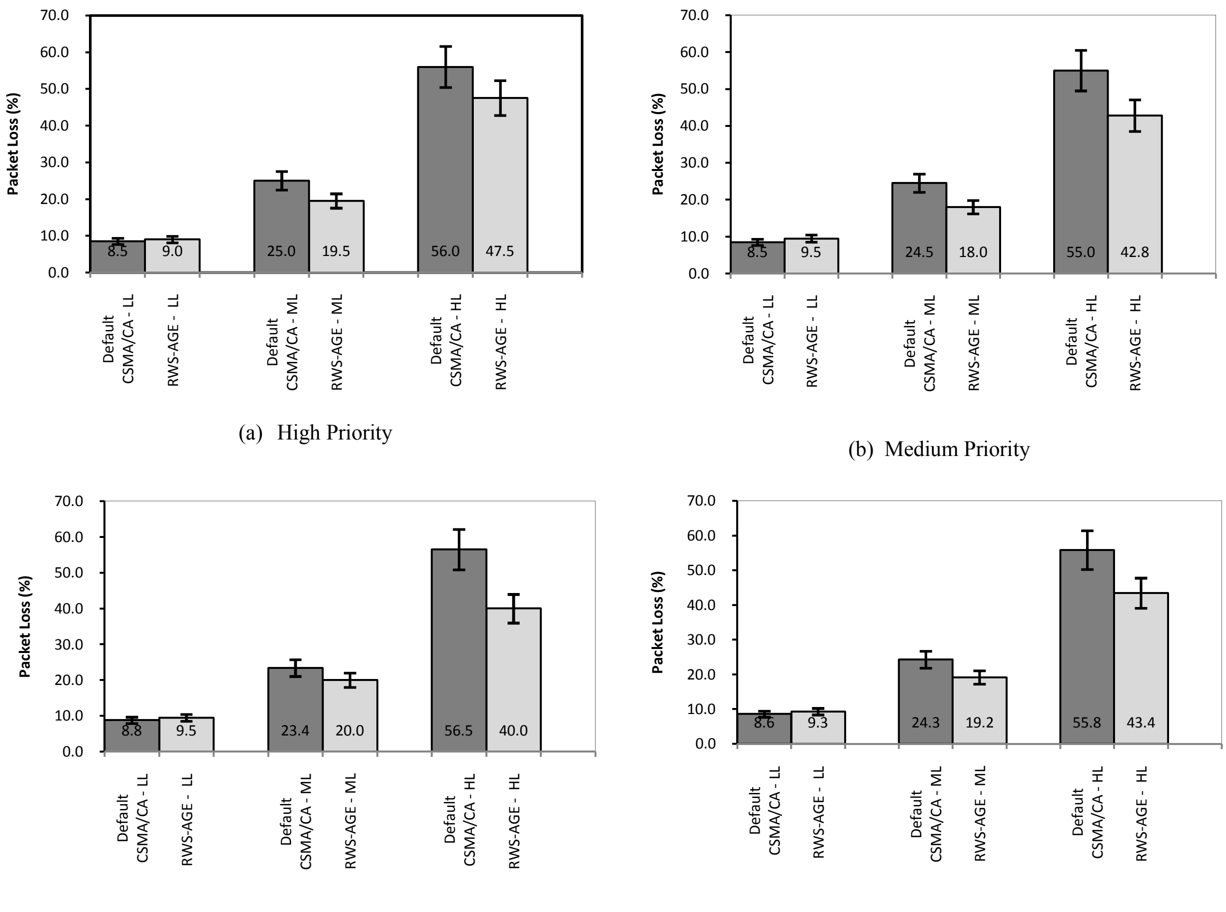


(c) Low Priority



(d) Average Packet Loss

Figure 6.7: Packet loss in test topology 1 (square grid).



(a) High Priority

(b) Medium Priority

(c) Low Priority

(d) Average Packet Loss

Figure 6.8: Packet loss in test topology 2 (line topology).

Chapter 7

Analytical Model

7.1 Introduction

CSMA/CA has two access techniques, namely a basic mechanism and the Request to Send/-Clear to Send (RTS/CTS) access mechanism. With the basic mechanism, the transmitter sends a packet, and the receiver acknowledges if it received the packet by sending an acknowledgement (ACK) message. With RTS/CTS, the channel is reserved before transmission. This chapter presents a model to predict the end-to-end delay of which the proposed end-to-end delay model is made up of the waiting time as well as the service time at each hop to reach the destination. The model used to calculate the end-to-end delay consists of three sub-models which are presented in turn. First, an absorbing state Markov chain model is developed to determine the expected number of re-transmissions at each hop. Secondly, the access delay model is derived. Lastly, we derive the expected end-to-end delay by using the values obtained from the access delay model and the expected re-transmissions models. This analytical model is suitable for modeling of general strategies that first select a packet for transmission and then perform back-off contention in multi-hop networks. Examples of such strategies are the basic distributed coordination function (DCF) access mechanism which does not differentiate between data and for data differentiated strategies such as RWS. The model is tested with DCF and with RWS. First-order Markov models work on the philosophy that the performance of the current state does not depend on the history of the previous states and can only be used for scheduling strategies whose operation do not depend on history. The proposed model conforms to the concept of a Markov model. The number of re-transmissions plays a critical role in the achievable end-to-end delay in multi-hop networks as well as the bandwidth consumption in a network. To the best of our knowledge, this is the first analytical model to use this approach.

7.2 Related Work

The Bianchi model proposed for the distributed coordination function (DCF) was one of the first analytical models to predict the performance in CSMA/CA [130]. The Bianchi model computes the IEEE 802.11 DCF saturated throughput by making assumptions of all nodes within the transmission range for a single-hop network, and uses ideal channel conditions. The model also considers the RTS/CTS access mechanisms and proposes a Markov chain approach model to model the binary back-off process [130].

Over the last decade, the Bianchi model has become the foundation for many other models such as analytical models for the priority based EDCA strategy [131–135]. Most of these models fall in either of two cases, namely saturation or non-saturation load conditions. By saturation, we mean that the node always has a packet to send and by non-saturation that the node does not always have a packet to send. In the Bianchi's model, the countdown timer for the back-off does not stop when the channel becomes busy. In [132], Xiao built on the work of Bianchi to develop a model to analyze the contention window size differentiation for the different priority queues in EDCA, but assumes equal arbitration inter-frame space (AIFS) periods for all traffic classes. In [131], an analytical model for EDCA throughput is proposed which considers collision probabilities for both cases, with and without using a virtual collision handler (VCH). In [133], the performance of EDCA is analyzed, based on both AIFS and retry limits for the contention window range, building on the work of [132]. In [134], expressions for the non-saturation throughput in EDCA are developed. In [136], an analytical model for both saturated and non-saturated throughput and end-to-end delay of the different traffic classes is proposed. In [135] a saturation throughput model has been developed.

In [137], the authors propose a model to analyse the throughput of EDCA in multi-hop networks. The model does not compare the analytical results with any simulations or testbed implementation results. The work takes into account non-saturation traffic conditions and hidden node problems by decomposing the problem in two models, based on a Markov chain. One model is for the node and the other is for the channel conditions. In [138], an analytical model for queuing delay in EDCA is presented. The model is analyzed for single-hop scenarios.

A three dimensional Markov chain model for the back-off operation is proposed in [139]. They derive the throughput for saturation conditions and do not consider the virtual collision mechanism. The authors of [137] propose two separate Markov chain models to model the different priority queues and the channel state conditions. They analyze the throughput and access delay in multi-hop networks. The authors in [140] also propose a three dimensional Markov chain model for the back-off operation for single-hop networks. Their model is an extension of Bianchi's model for saturated and non-saturated traffic. Other models that calculate the saturated throughput by extending Bianchi's model in single-hop networks include [141–143] and [133]. Non-saturated throughput is calculated in the work by [134] which is also an extension of Bianchi's model in single-hop networks. A novel high performance EDCA approach called H-EDCA to partition the collision domain of different classes of traffic based on Bianchi's model is proposed in [144]. A different approach than using the Markov chain has been applied by the authors in [145] by using hierarchical stochastic activity networks (HSAN), which is a stochastic Petri network to calculate the throughput. The work in [146] also uses Markov chains to model the back-off mechanism and is an extension of Xiao's model. They consider not only saturated traffic, but also non-saturated traffic for throughput and delay calculations. They also consider access delay in their model. A mean values analysis approach is used in [147] to calculate the saturated throughput for single-hop networks. The model considers the change of the CW size and AIFS. All these models have been applied for single-hop networks. A model for collision probability, throughput and access delay for both saturated and non-saturated delay for single-hop networks based on Bianchi's model is proposed in [148] for EDCA. They also use Pareto optimal pairs in their work for the number of stations and for different parameter sets and loads. A survey of DCF and EDCA models applied to single-hop networks is presented in [149].

Most of these Markov models are Discrete Time Markov Chain (DTMC) Models. The bi-dimensional Markov model has become a frequently used tool used to analyze the performance in CSMA/CA considering the back-off duration. The majority of this analytical work focuses on calculating the throughput and the mean delay by only considering the delay on the medium and not the queuing time in the node. Numerous models exist which are designed and applied to single-hop networks and not multi-hop networks. Interference plays a significant role on the performance in multi-hop networks as shown in [51–53] and [19] as the carrier sensing range is always greater than the transmission range in cases of overlapping collision domains [52]. A problem known as adjacent channel interference (ACI) exists in multi-hop networks where "bleeding over" takes place. This causes sensing from outside of its transmission range. Therefore, whenever a node within the interference range transmits, all other nodes within this range have to wait [53]. This interference is called co-channel interference and will interfere with the transmission of its neighbours using the same channel as if they were within the same interference range of each other and affect the capacity of the network [19]. Most models do not consider the re-transmission limit in their approach and do not calculate the estimated number of transmissions that take place. In summary, the aspects not covered by existing work are multi-hop networks with re-transmission calculations, access delay and capacity degrading with an increase in the number of nodes in the network. The advantage of our proposed model is that it considers these values in multi-hop networks.

7.3 Absorbing Markov Chain Modeling

We use an absorbing state Markov chain model to predict the expected number of re-transmissions at each hop in a network. In this section we present a brief overview of Markov chain theory. A Markov chain is a very popular stochastic model used to model dynamic systems that change its states over time. They can be classified as either being discrete time Markov chain (DTMC) or continuous time Markov chain (CTMC) [150].

A stochastic process X_n where discrete time $n \in N$ where $N = 0, 1, 2, \cdots$ is a state from a set of possible states for the system is known as a DTMC. X_n presents the state of the chain at n. Markov chains follow the Markov property which states that the behavior of the next state depends only on state the system is at present and not the past states [150,151].

In a Markov chain, a state can transit to the next state at time n. A Markov chain is known to be absorbing if it has an absorbing state such that once this state is entered, the model cannot exit this state. An essential feature of absorbing Markov chains (AMC) is that eventually an absorbing state is entered [150,152].

To solve absorbing Markov chains, the following steps are followed [152]:

1. The transition matrix is written in standard form. P is the transition matrix of a DTMC such that p_{st} is the probability of a transition from state s to state t. For an absorbing Markov chain, all the absorbing states are written such that they precede the non-absorbing states. The general standard canonical form matrix P is given as:

$$P = \begin{array}{cc} & \begin{array}{cc} \text{Absorbing} & \text{Non - absorbing} \end{array} \\ \begin{array}{c} \text{Absorbing} \\ \text{Non - absorbing} \end{array} & \left[\begin{array}{c|c} I & 0 \\ \hline R & Q \end{array} \right] \end{array}$$

P now has four sub-matrices, where I is the identity matrix, 0 is the zero matrix, R is the matrix of transition probability from a non-absorbing state to an absorbing state and Q is the matrix for transition probability from a non-absorbing state to a non-absorbing state.

P now has four sub-matrices, where:

a) I is a square identity matrix of size equal to the number of absorbing states for the number of rows and also the number of absorbing states for the number of columns.

b) 0 is a rectangular zero matrix of size equal to the number of absorbing states for the number of rows and the number of non-absorbing states for the number of columns.

c) R is the rectangular matrix of transition probability from a non-absorbing state to an absorbing state of size equal to the number of non-absorbing states for the number of rows and the number of absorbing states for the number of columns.

d) Q is a square matrix for transition probability from a non-absorbing state to a non-absorbing state of size equal to the number of non-absorbing states for the number of rows and also the number of non-absorbing states for the number of columns.

2. To determine the limiting matrix steady state (or the long run behavior of an absorbing Markov chain), we multiple P by itself continuously.

$$P^2 = \begin{bmatrix} I & 0 \\ R & Q \end{bmatrix} \begin{bmatrix} I & 0 \\ R & Q \end{bmatrix} = \begin{bmatrix} I & 0 \\ R+QR & Q^2 \end{bmatrix}$$

$$P^3 = \begin{bmatrix} I & 0 \\ R+QR & Q^2 \end{bmatrix} \begin{bmatrix} I & 0 \\ R & Q \end{bmatrix} = \begin{bmatrix} I & 0 \\ R+QR+Q^2R & Q^3 \end{bmatrix}$$

$$P^t = \begin{bmatrix} I & 0 \\ (I+Q+Q^2+ \dots\dots + Q^{t-1})R & Q^t \end{bmatrix}$$

As $t \longrightarrow \infty$, then $Q_t \longrightarrow 0$. The limiting matrix form now obtained has the form:

$$\overline{P} = \begin{array}{cc} & \begin{array}{cc} \text{Absorbing} & \text{Non - absorbing} \end{array} \\ \begin{array}{c} \text{Absorbing} \\ \text{Non - absorbing} \end{array} & \left[\begin{array}{c|c} I & 0 \\ \hline FR & 0 \end{array}\right] \end{array}$$

The system will move to some absorbing state. The limiting matrix is a simplified notation of multiplying P with itself until infinity. The fundamental matrix (F) is calculated as:

$$F = (I - Q)^{-1} \tag{7.3.1}$$

To determine the limiting matrix, FR must be calculated and then written in the form above.

3. The number of steps that it takes to reach the absorbing node(s) is calculated by summing each row in the fundamental matrix (F) to give the expected numbers of periods spent in each non-absorbing state before reaching the absorbing state. This is shown below as t. The summation of each row can easily be derived by multiplying the F matrix with a column matrix whose entries are all one.

$$t = Fc \tag{7.3.2}$$

   Where c is a column matrix whose entries are all one.

4. The following equation is used to determine the probability of entering an absorbing state given the current state.

$$B = FR \tag{7.3.3}$$

   $B_{y,z}$ is the probability of being absorbed in the absorbing state z from a transient state y. In B, y are the row elements representing the non-absorbing states and z are the column elements representing the absorbing states.

7.4 Assumptions and Network

This section presents the assumptions made to develop the analytical model, as well as the multi-hop networks used for testing of the model and to obtain the results in section 7.7.

A node is made up of arriving packets, packets being serviced and packets queued. A model of a single queue node is shown in figure 7.1.

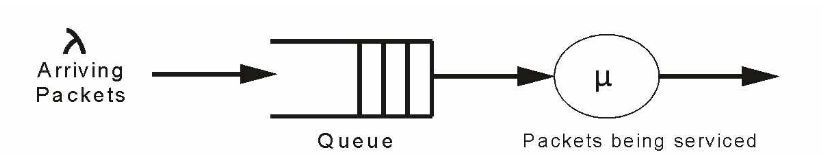


Figure 7.1: Model of a node with a single queue.

The following assumptions are made:

1. The packets for the different priority classes are of equal length. As stated in section 4.6, in many telemetry networks, packet sizes are usually between 60 bytes and 600 bytes and smaller packets have a lower probability of collision as they are less prone to collisions [114]. Therefore, in the model, the packet size is not taken into account.

2. The arrival rate is assumed to be a random Poisson distribution as the arrival events occur independently.

3. Each queue in each node follows the M/M/1 queuing principles as the arrival rates are assumed to be a Poisson distribution and the departure rates are assumed to have an exponential distribution. Both these distributions are memoryless. Both these distributions are memory-less and thus we use the M/M/1 queue model. The 1 is used as we use a single channel for transmission.

4. The optional RTS/CTS mechanism in IEEE 802.11 standard is not used, as the mechanism under study is the basic mechanism.

5. The channel is in ideal condition. This is to say that the model assumes there are no channel errors, no capture effects and no hidden terminal problems.

6. All nodes are in saturation. This is to say that it is assumed that every node always has data to transmit in its buffer.

7. The system is slotted. This is to say that a node is only allowed to send data at the beginning of the time slot. The count down for the back-off takes place in discrete time step intervals equal to the time slot. This condition makes it possible to model the system as a DTMC model.

8. The collision probability is constant for a given traffic load depending on the network size and priority data class.

9. The queuing system is open, meaning packets can enter and leave the queue in a node.

10. The data priority queues in each node have infinite length and no packets are dropped due to congestion.

11. The TXOP limit is not used. If TXOP bursting is used where multiple packets are allowed to be transmitted when the node gains access to the channel, than the queue will follow a G/G/1 queuing system and the M/M/1 equations will not hold.

In this model, only the important parts of the MAC layer scheduling strategies are modeled while the less important parts that are not under study are simplified or omitted. The important parts are the scheduling strategy operation, the selection of different priority data, transmission probability based on the load level, queue waiting time and collisions. These less important parts are information of other layers such as the application, transport, network and physical layers and their overheads.

Different multi-hop network sizes from a 1 hop network size to a 5 hop network size as shown in figure 7.2 are used to obtain the results in the analysis of the model presented in the next section. A maximum network size of 5 is used for two reasons. The first being that it is rare to have networks with data having to be transmitted over a large number of hops. Secondly, the model holds under stability conditions. A large number of hops for transmission perhaps could make the system unstable. The transmission range for each node is shown as dotted lines. The transmission range for each node is shown as dotted lines. The interference range is larger than the transmission range as the strategy uses single-radio single-channel technology [52,153]. Another node within the interference range that wants to send data will detect the channel as being busy if a transmission is taking place by a node within the transmission range. The end nodes (source nodes) send data to the destination nodes for the different network sizes as shown in figure 7.2. All the intermediate nodes forward data. The distance between the nodes has an effect on the propagation delay to transmit the packet between the nodes as well as the bit error rate (BER). The transmission range of each node in the networks is such that only one hop transmission can take place.

7.5 End-to-End Delay Analytical Model

The end-to-end delay is defined as the time that elapses from the time the packet is sent to the time that it successfully reaches the destination. The end-to-end delay is made up of the total

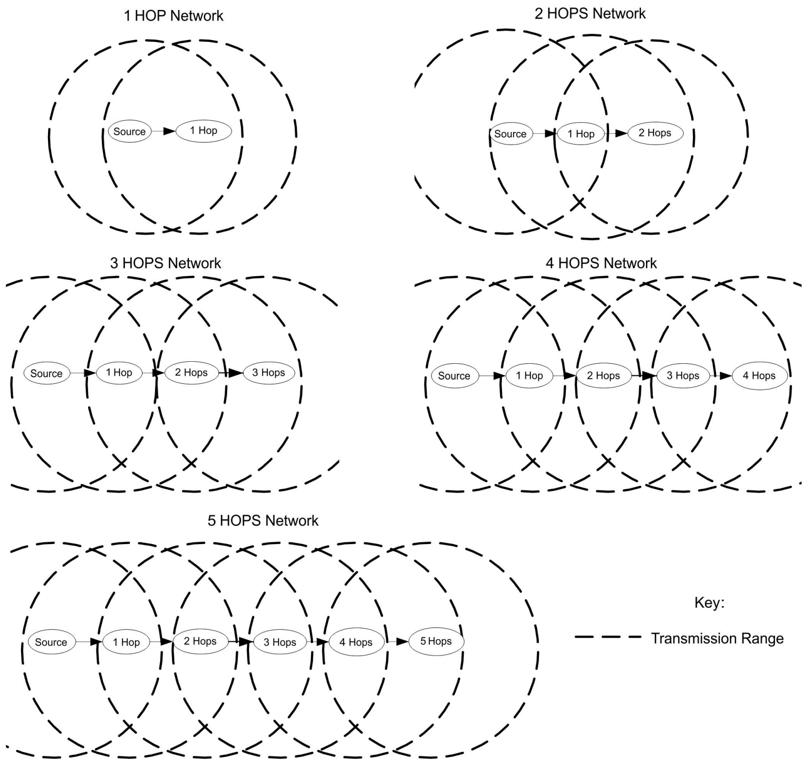


Figure 7.2: Multi-hop networks under study.

service times and waiting times in the queue at each hop link for a packet to reach the destination from the source node. In other words, the true sojourn time is the sum of these basic access times with the queuing times at each node at each hop. The waiting times at each node are made up of the access delay time, the arbitration interframe spacing (AIFS) for a multi-queue system or DCF Interframe Space (DIFS) for a single queue system and the back-off time which depends on the contention window (CW) size. The service time on each link is made up of the time to transmit the header of the packet, the time to transmit the payload of the packet, the short interframe space (SIFS) period, time to transmit the acknowledgement (ACK) message, propagation delay, ACK-timeout period in the event that no ACK is received, and the number of transmissions at each hop link. These parameters are illustrated in figure 7.3 for each hop link. The medium access delay equations are also presented in [39], [131] and [134]. In this section we develop a model to calculate the end-to-end delay for a single queue SBC strategy such as DCF as well as for a multi-queue SBC strategy such as RWS.

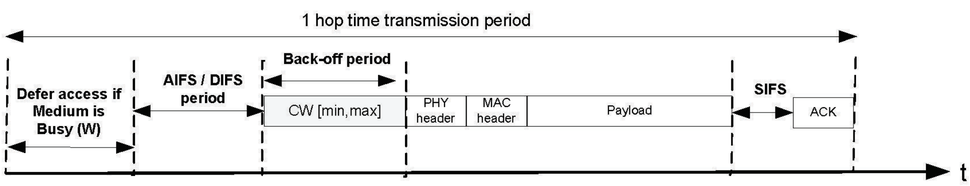


Figure 7.3: Timing Diagram for sojourn time at each hop link.

7.5.1 Single Queue Strategy

DCF uses a single queue in each node with data treated in a first-in-first out (FIFO) manner. With DCF, the end-to-end delay for a successful transmission on the h_{th} hop link (TS_h) is given by:

$$TS_h = DIFS + BO + SIFS + ACK + W_h + PropDelay + \frac{L}{R_t} \tag{7.5.1}$$

Where $DIFS$ is the DIFS duration; $SIFS$ is the SIFS time period; ACK is time to transmit back an acknowledgement; W_h is the access delay time at the h_{th} hop; $PropDelay$ is the propagation delay time which is the time taken to transmit a signal based on the distance between the nodes; L is the size of the packet including the header and payload; BO is the back-off duration which depends on the CW value selected; and R_t is the average transmission rate on the medium.

If a collision takes place, the collision time is expressed as:

$$TC = DIFS + BO + SIFS + ACK_{TIMEOUT} + PropDelay + \frac{L}{R_t} \tag{7.5.2}$$

$$ACK_{TIMEOUT} = SIFS + ACK + PropDelay \tag{7.5.3}$$

Equations 7.5.2 and 7.5.3 are derived fronfact that if a node does not receive an acknowledgment from the receiver within a time period of ACKTIMEOUT, the sender assumes a collision occurred or the packet did not successful reach the destination. Therefore, another transmission attempt is made. The collision time in equation 7 therefore, takes into account the additional ACKTIMEOUT period before attempting another transmission attempt.

The end-to-end delay (D) over all the hop links takes into account the successful transmission time on each hop link, collision time and the number of re-transmissions as:

$$D = \sum_{h=1}^{H} (TS_h + NR_h * TC) \tag{7.5.4}$$

Where NR_h is the number of re-transmission at the h_{th} hop link.

Section 7.5.1.1 now explains how the expected number of re-transmissions at each hop is calculated. Section 7.5.1.2 explains how the access delay time is calculated. Section 7.5.1.3 explains how the stability of the system can be calculated.

7.5.1.1 Expected number of Re-transmissions

To calculate the expected number of re-transmissions in a multi-hop path network, we modeled the system as an absorbing state DTMC. The notation used to represent the states is: hop number, transmission number. States $1,8$; $2,8$; $3,8$; $\cdots$ up to $N,8$ and the destination state are all made absorbing states. Since only 7 re-transmissions are allowed, the 8_{th} transmission attempt represents an unsuccessful transmission of a packet where the packet is dropped. A transition to the next hop node depends on the probability of success on the channel. A transition to the next transmission attempt state at the same hop node depends on the probability of not being successful. A N-hop network is shown in figure 7.4. Figure 7.5 shows the possible transitions of non-absorbing states.

Limiting matrices for the different network sizes are derived. For a 1 hop network, the transition matrix written in standard form as explained in section 7.3 with the absorbing states written ahead of the non-absorbing states, is:

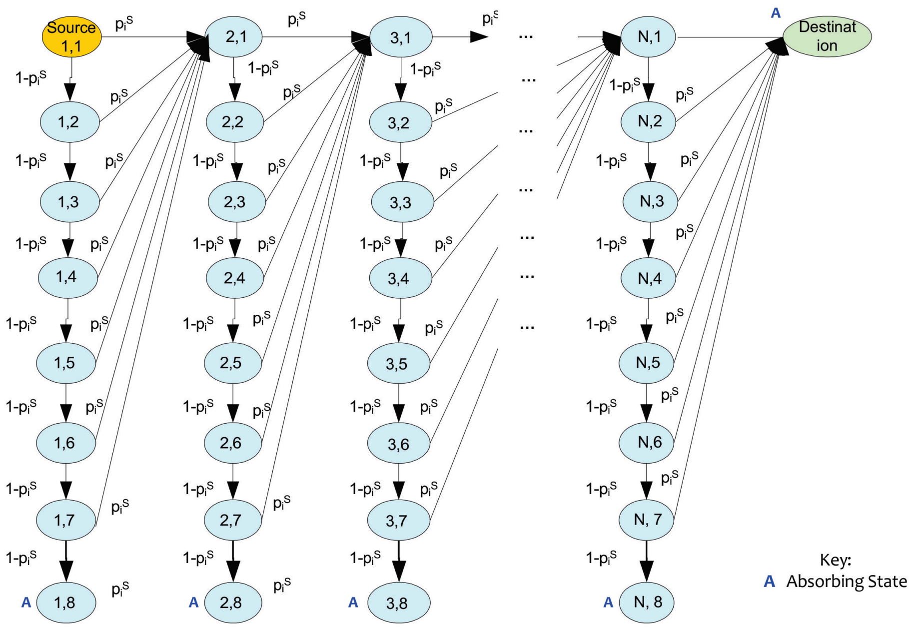
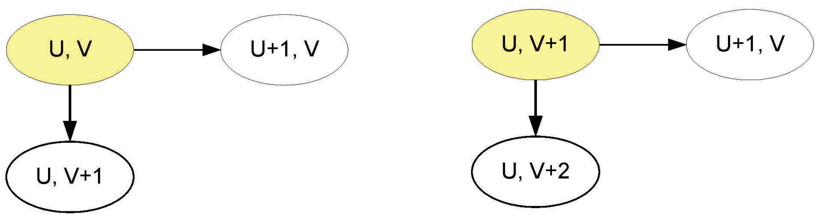


Figure 7.4: The absorbing markov chain model for a multi-hop network.



(a) A non-absorbing current node state can either move to the next hop node state (on the right) or enter the next transmission attempt state if a failure occurs (state to the bottom).

(b) A non-absorbing current next transmission attempt node state can either move to the next hop node state (on the right) or enter the next transmission attempt state if a failure occurs (state to the bottom).

Figure 7.5: Non-absorbing state transitions.

$$\overline{P}_{1hopnetwork} = \begin{array}{c|c} & \begin{array}{cc|ccccccc} \multicolumn{2}{c|}{\text{Absorbing states}} & \multicolumn{7}{c}{\text{Non-absorbing states}} \\ 1{,}8 & Destination & 1{,}1 & 1{,}2 & 1{,}3 & 1{,}4 & 1{,}5 & 1{,}6 & 1{,}7 \end{array} \\ \begin{array}{r} \text{Absorbing states} \; 1{,}8 \\ Destination \\ \text{Non-absorbing states} \; 1{,}1 \\ 1{,}2 \\ 1{,}3 \\ 1{,}4 \\ 1{,}5 \\ 1{,}6 \\ 1{,}7 \end{array} & \left(\begin{array}{cc|ccccccc} 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ \hline 0 & P_s & 0 & 1-P_s & 0 & 0 & 0 & 0 & 0 \\ 0 & P_s & 0 & 0 & 1-P_s & 0 & 0 & 0 & 0 \\ 0 & P_s & 0 & 0 & 0 & 1-P_s & 0 & 0 & 0 \\ 0 & P_s & 0 & 0 & 0 & 0 & 1-P_s & 0 & 0 \\ 0 & P_s & 0 & 0 & 0 & 0 & 0 & 1-P_s & 0 \\ 0 & P_s & 0 & 0 & 0 & 0 & 0 & 0 & 1-P_s \\ 1-P_s & P_s & 0 & 0 & 0 & 0 & 0 & 0 & 0 \end{array}\right) \end{array}$$

These matrices are then used to determine the expected number of transmissions between each hop to reach the destination and the total number of steps before it reaches the destination node can be calculated with the application of equation 7.3.2. (the number of steps that it takes to reach the absorbing node). The probability of reaching a certain absorbing state is calculated using equation 7.3.3. (the probability that if a system is in a particular state, the probability that it will reach the absorbed state).

To determine the probability of successful transmission on the channel (value needed in the limiting matrices), we first derive channel statistical probabilities for collision on the channel, the probability of error on the channel and use these probabilities to calculate the probability of successful transmission on the channel.

The probabilities are estimated as follows:

1. **Probability of Collision on the channel:** The probability of collision in a time slot is dependent on the transmission attempt probability and number of nodes that can not transmit if any one of the other nodes is transmitting. The probability of collision is calculated based on the fact that a collision takes place when one node transmits and any of the N-1 remaining nodes also transmit a packet in the same time slot. The transmission attempt probability of a node is inversely proportional to its contention window size under saturation [154–158]. We assume the value of CW selected as CW_{min} when the first collision takes place. The probability of collision neglects multiple re-transmissions as the probability of success needed in the model is based on the first attempt. The transmission attempt is statistically independent of the other.

   The transmission attempt probability (τ) can be expressed as:

   $$\tau = \frac{1}{CW_{min}} \tag{7.5.5}$$

   The system is backlogged if the queue is not empty for any queue as stated in [159]. The transmission probability increases with an increase in load.This happens as a result of queues with data count down when the channel is detected as being idle and freeze when a transmission takes place. With a higher load, there is a higher chance of more packets already having counted down for the back-off and thus, increasing the transmission attempt probability. This can be taken care of in the denominator taking the arrival and departure rate of the nodes in the transmission range. For networks with nodes in the interference range, bleeding and overlapping of transmission ranges causes nodes trying to transmit to hear other communications beyond the transmission range. This happens if SRSC technology is used and all nodes are configured to the same channel frequency. A high signal in one region can cause neighbours to detect the high signal and thus prevent them from transmitting. Thus, this causes their transmission to be delayed further. The number of nodes within the interference range that prevents concurrent transmission depends on the size of the network. The new transmission attempt probability taking load and the nodes in the interference range now becomes:

   $$\tau = \frac{1}{(CW_{min})(1-\rho)} \tag{7.5.6}$$

   where

   $$\rho = \frac{\lambda}{\mu} \tag{7.5.7}$$

   The simplified derived equation for collision probability in a slot time now becomes:

   $$P_c = 1 - (1-\tau)^{N-1} \tag{7.5.8}$$

The system holds for the following conditions:
$0 \leqslant \rho < 1$
$0 \leqslant \tau < 1$

Where N is the number of nodes in the interference range, ρ is the traffic density (also known as the utilization factor), μ is the departure rate and λ is the arrival rate.

2. **Probability of error in transmission:** Bit Error Rate (BER) also plays a significant role in the successful receiving of a packet. Packets with errors also have to be re-transmitted. Packet size has a great impact on the performance of the system. Larger packet size will result in higher collisions. The probability of error is calculated based on BER and packet length (L) as:
$$P_{error} = 1 - (1 - BER)^L \tag{7.5.9}$$
We assume ideal channel conditions and therefore BER = 0.

3. **Probability of successful transmission:** The overall probability of successful transmission over the channel is calculated as:
$$P_s = 1 - P_c - P_{error} \tag{7.5.10}$$
The calculated probability of successful transmission is used for the absorbing Markov chain model presented in figure 7.4. Equation 7.3.2 is used to determine the expected number of transmissions between each hop to reach the destination.

4. **Average CW size:** We calculate the average back-off CW size using the work of [160]. The probability of collision on the channel is P_c and probability of no collision is $1 - P_c$. These values are used in a renewal reward process to calculate the approximate average back-off CW as a geometric distribution counting the results up to the first success considering re-transmissions:
$$CW_{avgbackoff} = (1-P_c)\frac{CW_{min}}{2} + P_c(1-P_c)\frac{2CW_{min}}{2} + ... + P_c^M(1-P_c)\frac{2^M CW_{min}}{2} + P_c^{M+1}\frac{2^M CW_{min}}{2} \tag{7.5.11}$$
Where M is the maximum number of allowed exponential increase for the CW size and CW_{min} is the minimum CW range size value. It gives us the expected number CW back-off re-transmission sizes between the renewal events. With a probability of $1 - P_c$ the transmission is successful and the random number generated is assumed to be the mid value between 0 and CW_{min}. The second value in the equation is when the first transmission fails and the second is successful. The contention window increases in size as well. The rest of the terms allow for up to M re-transmissions.
The obtained value for the number of transmissions can also be verified by calculating the average CW size from equation 7.5.11 and determining the range in which this value falls in. The CW_{min} is incremented exponentially after every collision until it reaches CW_{max}. After it reaches CW_{max}, it stays constant until a success transmission or until the packet is dropped by reaching the maximum retransmission limit.

7.5.1.2 Access Delay Time

The queue in a node with DCF is assumed to follow the principle of a M/M/1 queue. In [161], the DCF in multi-hop networks also model the queues in a node using M/M/1 queue. In [162], EDCA has been modeled as M/M/1. It is an M/M/1 queue as the arrival rate follows a Poisson distribution with a fixed arrival rate for constant bit rate (CBR) data and the events occur independently; the departure rate is exponential as the waiting time between events in unknown and random; and the system has 1 channel for service of the packets. The departures from one node feed as arrivals to the next hop node. The access delay time (W) which is basically the time the packet waits in the queue is derived [163] as:

$$W = \frac{\rho}{\mu - \lambda} \tag{7.5.12}$$

7.5.1.3 Determining Network Stability

The number of packets in the node system gives a good indication if the system is in saturation or not. The number of packets in the system for an M/M/1 system can be calculated as [163]:

$$Pkt_s = \frac{\rho}{1 - \rho} \tag{7.5.13}$$

The utilization equation presented in equation 7.5.7 indicates the stability of a system. If the utilization is below 1, the system is known to be stable. If the calculated value is greater than 1, the system becomes unstable [163]. To determine if the system will be stable for the evaluated arrival rate, for different network sizes, the utilization can be estimated considering each link in the interference range of the network under study.

7.5.2 Multi-Queue Strategy

In section 7.5.1 we derived the end-to-end model for a single queue multi-hop system. In this section we derive a model for a multi-queue system. We use the RWS scheduling strategy. Data packets for each priority are of equal size. The system is a non pre-emptive queuing system. For a pre-emptive priority queuing system, the on-going service is halted when a higher priority class data arrives. For a non pre-emptive priority system, the current service is not halted even if higher priority data arrives.

With RWS, the end-to-end delay for a successful transmission on the h_{th} hop link for the j_{th} priority class ($j = 1, 2, \cdots, J$) where 1 is the highest priority queue and J represents the lowest priority queue) is given by:

$$TS_{j,h} = AIFS(j) + BO(j) + SIFS + ACK + W_{j,h} + PropDelay + \frac{L}{R_j} \tag{7.5.14}$$

Where $AIFS(j)$ is the AIFS duration for j_{th} priority class; $W_{j,h}$ is the access delay time at the h_{th} hop for the j_{th} priority class; BO(j) is the back-off duration for the j_{th} priority class; and R_j is the average transmission rate for priority class j_{th} on the medium.

If a collision takes place, the collision time is expressed as:

$$TC = AIFS(j) + BO(j) + SIFS + ACK_{TIMEOUT} + PropDelay + \frac{L}{R_j} \tag{7.5.15}$$

The end-to-end delay for the j_{th} priority queue (D_j) over all the hop links takes into account the successful transmission time on each hop link, collision time and the number of re-transmissions.

$$D_j = \sum_{h=1}^{H}(TS_{j,h} + NR_{j,h} * TC) \tag{7.5.16}$$

where $NR_{j,h}$ is the number of re-transmissions at the h_{th} hop link for the j_{th} priority class.

To calculate the expected number of re-transmissions at each hop for each priority class, the same AMC model as in section 7.5.1.1 is used, except that equations are derived to consider the multiple queues as given in section 7.5.2.1. Section 7.5.2.2 explains how the access delay time in RWS for each priority queue is calculated.

7.5.2.1 Channel Probabilities using multi-queue

The collision probability on the channel under conditions of stability considering possible collisions due to packets from any one of the j_{th} priority classes from any of the other nodes is:

$$P_c = 1 - \prod_{j=1}^{J}(1 - \tau_j)^{N-1} \tag{7.5.17}$$

Where τ_j is the transmission probability of the j_{th} priority class data.

The probability of successful transmission is calculated as given in equation 7.5.10.

7.5.2.2 Access Delay using multi-queue

In a multi-queue system with J priority classes ($j = 1, 2, \cdots, J$), the arrival rates of the different classes are $\lambda_1, \lambda_2, \cdots, \lambda_J$. The mean and second moment of the service time of the different classes are $\overline{x_j}$ and $\overline{x}_j^2$. In this work, the derived mean results of the Pollaczek-Khinchine type for M/G/1 priority system is used and extended to convert the system to M/M/1 as per the rule in [163]. The data packets from the different queues are served in the order they arrive at the particular queue.

In [163], the M/G/1 non-preemptive priority queue model is presented. The model is simplified in our work to transmit data according to the assigned transmission probabilities from the different queues for the RWS scheduling strategy. The M/G/1 model is not suitable for our system as the departure rate in our system is assumed to follow an exponential distribution for a discrete probability distribution that assumes an estimated output probability scheduling from each queue in the scenario when the system is saturated. The departures from the current node feed as input to the next hop node and the arrival rates are totally independent. In M/G/1 model, the residual service time is therefore, modified to be exponential and the service time is assumed to be exponential as the next queue for packet transmission is chosen randomly with fixed rates. This changes the system to become an M/M/1 non-preemptive system priority system. An M/G/1 model is a semi-Markovian queuing system and is solved with techniques like imbedded Markov-chain and residual service time are used for solving M/G/1 problems. The imbedded process looks at the queue behavior at service completion while the residual service time method models the system from an arriving packets perspective. From the two methods, the residual method is simpler to use but only provides mean value results. The work in this model is based on the M/G/1 model on the Residual Service Time. The M/G/1 model is changed to an M/M/1 by changing the second moment to become $2\mu^{-2}$ for Residual Service Time according to the rules stated in [163] and [164].

The data packets are served by the same channel with a general service time distribution with a mean $\overline{x}_j$ and second moment $\overline{x}_j^2$ for data packets belonging to class j. Equations 7.5.18 to 7.5.23 have been extracted from [163] and how the waiting time is derived is explained briefly before we use the basis of this model to develop our access queuing delay model.

The total packet arrival rate for the different priority data is:

$$\lambda = \sum_{j=1}^{J} \lambda_j \tag{7.5.18}$$

The utilization of data packet from class j is given by:

$$\rho_j = \lambda \overline{x}_j \tag{7.5.19}$$

The average system service time and utilization become:

$$\overline{x} = \sum_{j=1}^{J} \frac{\lambda_j}{\lambda} \overline{x}_j \tag{7.5.20}$$

$$\rho = \sum_{j=1}^{J} \rho_j \tag{7.5.21}$$

The mean residual service time (R) is the weighted sum of all the residual service time for each priority class calculated in [163] as:

$$R = \sum_{j=1}^{J} \rho_j (\frac{\overline{x}_j^2}{2\overline{x}_j}) \tag{7.5.22}$$

The waiting time for a data packet for the for the n_{th} priority queue class that arrives at any of the different priority queues is made of the mean residual service time, the total service times of the data packets already in the same priority queue as well as service time when other queues are being serviced. The derived waiting time is:

$$W_n = \frac{R}{(1 - \rho_1 - \rho_2 - ... - \rho_{n-1})(1 - \rho_1 - \rho_2 - ... - \rho_n)} \tag{7.5.23}$$

Using the theory in [163], it is stated that if the service time is exponential, then the second moment becomes $2\mu_j^{-2}$ for Residual Service Time. Substituting this condition in equation 7.5.22, the new residual service time makes the model M/M/1. The new value of R now becomes:

$$R = \sum_{j=1}^{J} \lambda_j \overline{x}_j^2 = \sum_{j=1}^{J} \lambda_j \overline{\mu}_j^{-2} \tag{7.5.24}$$

Combining equations 7.2.23 and 7.2.24, we get the access delay time for the n_{th} priority queue in a node as:

$$W_n = \frac{\sum_{j=1}^{J} \lambda_j \overline{x}_j^2}{(1 - \sum_{j=1}^{n-1} \rho_j)(1 - \sum_{j=1}^{n} \rho_j)} \tag{7.5.25}$$

The access delay for the n_{th} priority queue at the h_{th} hop node can be written as:

$$W_{n,h} = \frac{\sum_{j=1}^{J} \lambda_{j,h} \overline{x}_{j,h}^2}{(1 - \sum_{j=1}^{n-1} \rho_{j,h})(1 - \sum_{j=1}^{n} \rho_{n,h})} \tag{7.5.26}$$

The channel utilization for the j_{th} class at the h_{th} hop can be written as:

$$\rho_{j,h} = \lambda_{j,h} \overline{x_{j,h}} \tag{7.5.27}$$

With the RWS scheduling strategy, a priority queue for a packet transmission is first chosen. This is done by assigning weights to each priority queue. After that a random number is generated. A priority queue in which the number falls in is chosen to transmit a packet. A packet is selected, and then the contention periods of AIFS and back-off are carried out. The analysis network model is shown in figure 7.2. Each node receives and forwards data to the destination node as shown in figure 7.6.

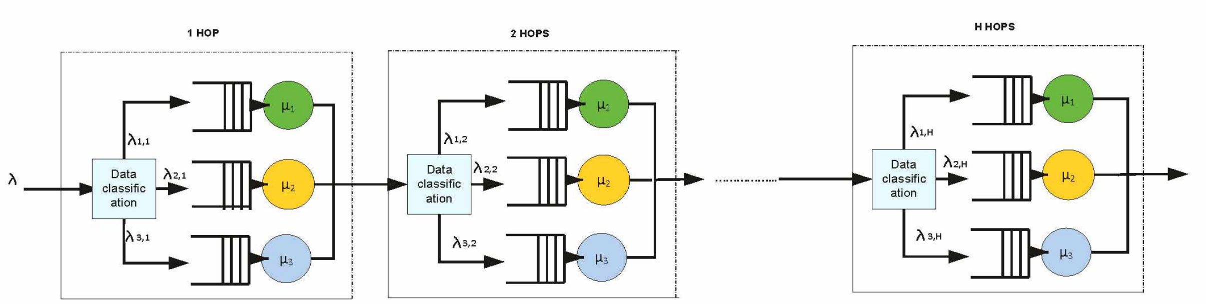


Figure 7.6: A multi-hop network using the RWS scheduling strategy.

The arrival rate for the j_{th} priority class at the h_{th} hop for the network under study in saturation can be expressed as:

$$\lambda_{j,h} = \lambda_h PW_j^{h-1} \tag{7.5.28}$$

Where PW_j is the scheduling weights assigned to the priority class in the RWS strategy.

Substituting equations 7.5.27 and 7.5.28 into equation 7.5.26, the access delay time in the queue for each priority queue at each hop node is derived as:

$$W_{n,h} = \frac{\sum_{j=1}^{J} \overline{x_{j,h}^2} \lambda_h PW_j^{h-1}}{(1 - \sum_{j=1}^{n-1} \overline{x_{j,h}} \lambda_{j,h})(1 - \sum_{j=1}^{n} \overline{x_{j,h}} \lambda_{j,h})} \tag{7.5.29}$$

Table 7.1: IEEE 802.11g parameter values used

Parameter	**Value**
Simulation Time	900s
Channel Parameters	
Path Loss Model	Free Space
slot time	$9\mu s$
Channel Data Bit Rate	$54Mbps$
SIFS	$10\mu s$
Basic Bit Rate	$6Mbps$
Propagation Delay	$1\mu s$
Packet Parameters	
PHY Preamble and Header	192 bits
MAC Header	272 bits
ACK	112 bits

Table 7.2: Default DCF and RWS strategy parameters used

	RWS Scheduling			**DCF Scheduling**
	High Priority (HP) Data	**Medium Priority (MP) Data**	**Low Priority (LP) Data**	
CWmin	7	15	31	31
CWmax	15	31	1023	1023
Retry Limit	7	7	7	7

7.6 Model Verification

To confirm the accuracy of the analytical model, we compare the prediction of the analytical model to simulation results which are obtained from OMNeT++. DCF has one queue in each node. For RWS, we implement three queues in each node. The parameters used in both simulations and the model are shown in tables 7.1 and 7.2. The results presented in this section are for different load scenarios using DCF and RWS as shown in table 7.3. Each node is configured with the IEEE 802.11g standard using 54 Mbps as the data rate and 6 Mbps as the basic rate. Acknowledgments are sent back at the basic rate and this is the rate at which packets leave the node for one queue system if it is the only node contending for the medium. Each simulation was run five times with different seed numbers for each run and the average values were used for the results. The maximum and minimum values obtained from the different seed runs were used to plot the simulation error bars. To obtain the numerical results of the analytical model, the equations were setup as functions in Matlab.

7.7 Results

This section presents the end-to-end delay results for both the analytical model and the simulations. The analytical model and simulated end-to-end delay results for the test case scenarios over the different hop network sizes are presented in figure 7.7 for DCF and in figure 7.8 for the three queue RWS scheduling test scenarios. Close correlation with the results is observed. It is observed that for the DCF case with one queue, the system becomes unstable for network sizes of greater than 4 hops for the load of 300 packets per seconds (pps). This is in agreement with the network utilization calculated using equation 7.5.7 of 1.1 which is greater than 1 and hence the system becomes unstable. The stability calculated for each network size and load is presented

Table 7.3: Test cases for different Schedule before contention strategies.

	Scheduling Strategy	Data Classes	Data rates (pps)	Total traffic (pps)	Network Utilization for the Different Network Sizes				
					1 Hop	2 Hops	3 Hops	4 Hops	5 Hops
Scenario 1	DCF	1	200	200	0.14	0.27	0.41	0.55	0.68
Scenario 2	DCF	1	300	300	0.22	0.44	0.66	0.88	1.10
Scenario 3	RWS	3 (HP, MP and LP)	50	150	0.11	0.22	0.33	0.44	0.55
Scenario 4	RWS	3 (HP, MP and LP)	100	300	0.22	0.44	0.66	0.88	1.10
Scenario 5	RWS	3 (HP, MP and LP)	150	450	0.33	0.66	0.99	1.33	1.66

in table 7.3. The model is only valid when the network is stable. The same instability effect is observed for the RWS cases with three queues for a data rate of 100pps for each queue after 4 hops. For test scenario 4 the 150pps case with RWS with a data rate of 150pps for a network size greater than 3 hops is also unstable. The error bars show the range in which the simulated results fall in with the use of different seeds. It is observed that with end-to-end delay, there is a small error bar range for the simulated results.

There are slight differences between the analytical model results and those of the simulated results for the end-to-end delay. In most cases, the simulated results are higher than the analytical model results. The percentage of absolute error between the simulated and analytical end-to-end delay results is calculated in table 7.4. The errors are all less than 17% with most cases being below 10%. Higher absolute errors are observed at 1 hop and 2 hop nodes as these nodes are not in saturation. Lower absolute errors are observed when the nodes are in saturation. Lower absolute errors are observed when the nodes are in saturation.The difference between the analytical model results and simulation results can be explained as follows: (1) The model does not take into account when two nodes from different collision domains transmit at the same time and a collision occurs. (2) The midvalue of the CW values are used for the back-off duration which is just an approximation. The actual value could be less or more than this value. (3) The model does not consider the period that the back-off freezes when another node is in communication. (4) The model does not consider information from the other layers such as the application, transport, network and physical layers and their overheads.

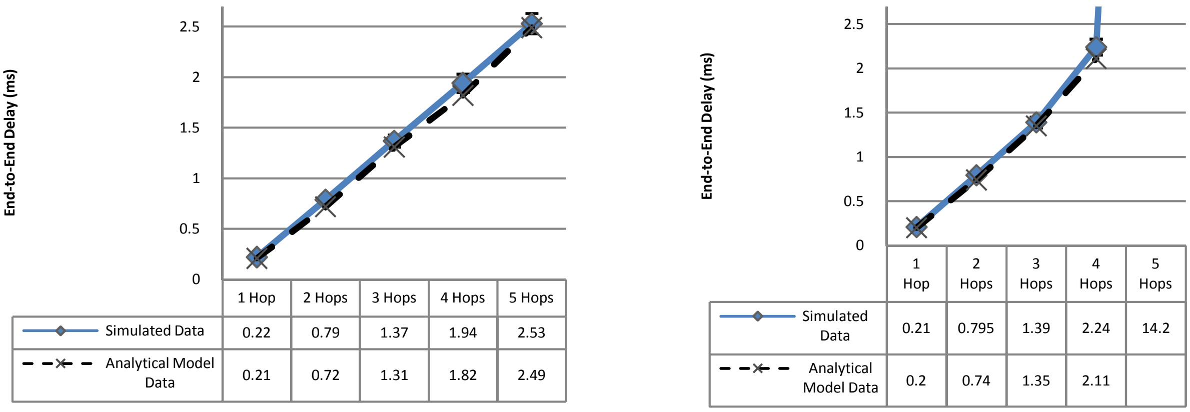
Collisions result in the need for re-transmissions which increases the end-to-end delay as data has to travel over multi-hops to reach its destination. The collision probability therefore, plays a critical role in the end-to-end delay achievable. The number of collisions increases with an increase in arrival rate and the size of the network for all priority data classes.

7.8 Conclusion

This chapter presented a statistical analytical model framework for the end-to-end delay calculations in a multi-hop network for schedule before contention MAC strategies. This model is applicable to both networks with nodes with a single queue or multiple queues for differentiated heterogeneous data. The analytical model applied Markov chain theory with absorbing states and queuing theory. M/M/1 queuing theory is used to represent the queues in a multi-hop network in order to derive the access queue waiting times. The model is applicable to single radio single channel networks and hence the number of servers used in the queuing theory is 1. The arrival and

Table 7.4: Percentage of absolute error between the simulated and analytical model results for the end-to-end delay.

	Scheduling Strategy	Data Classes	Data rates (pps)	Percentage of Error for End-to-End Delay (%)				
				1 Hop	2 Hops	3 Hops	4 Hops	5 Hops
Scenario 1	DCF	1	200	4.45	8.86	4.38	6.19	1.58
Scenario 2	DCF	1	300	4.76	6.92	2.88	5.80	
Scenario 3	RWS	3	50 HP	9.52	3.92	3.49	0.77	2.73
Scenario 3	RWS	3	50 MP	13.89	3.70	6.82	0.05	0.39
Scenario 3	RWS	3	50 LP	7.27	16.95	4.74	2.93	4.11
Scenario 4	RWS	3	100 HP	8.70	3.51	6.48	1.10	
Scenario 4	RWS	3	100 MP	8.33	2.77	4.61	4.62	
Scenario 4	RWS	3	100 LP	7.14	9.84	6.22	3.10	
Scenario 5	RWS	3	150 HP	12.00	7.69	5.71		
Scenario 5	RWS	3	150 MP	7.69	2.15	2.25		
Scenario 5	RWS	3	150 LP	10.71	2.36	6.87		

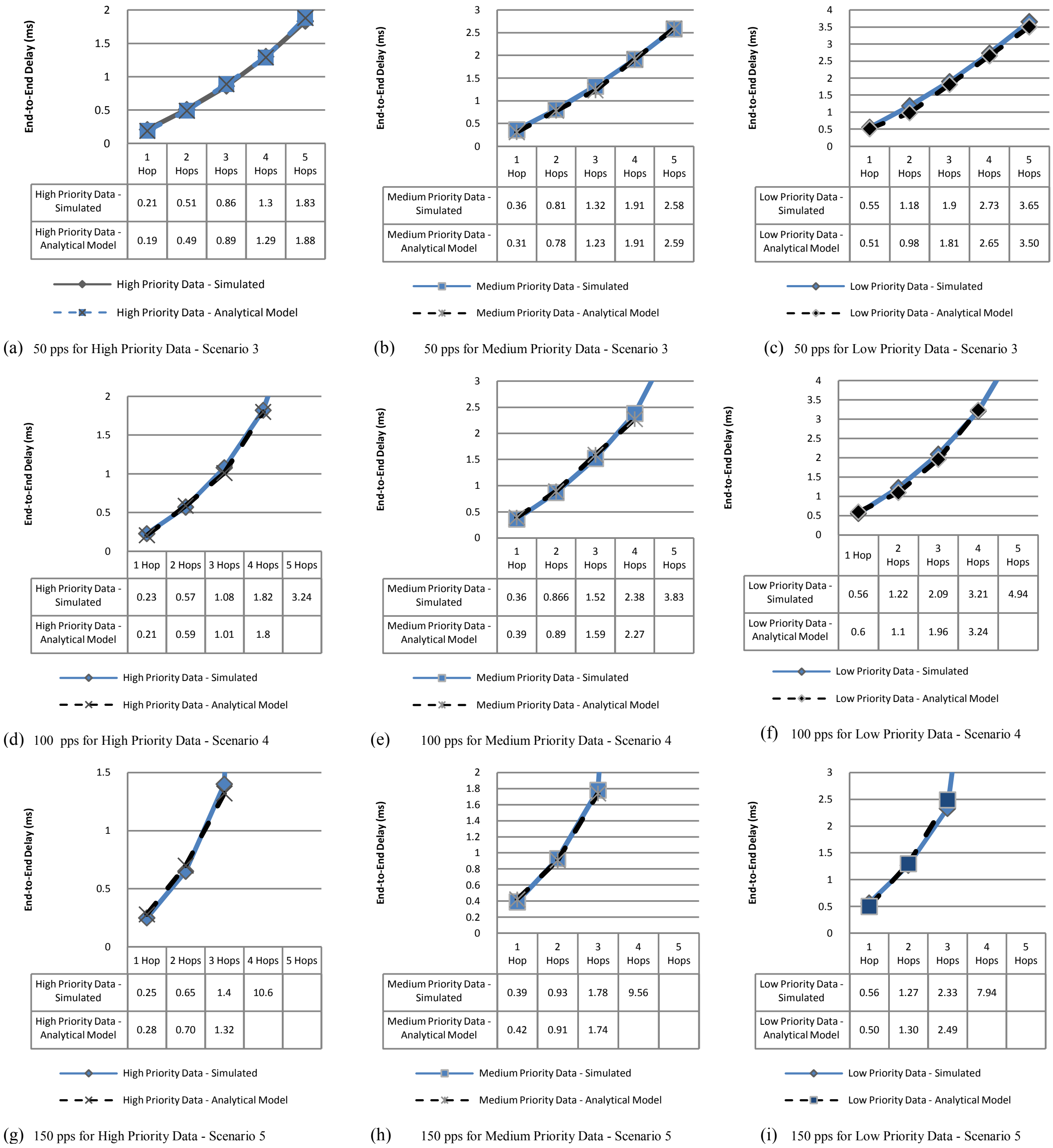


a) 200 pps – Scenario 1 b) 300 pps - Scenario 2

Figure 7.7: End-to-End Delay with the DCF scenarios.

departure rates from each queue are assumed to be Markovian. A Poisson random process, whose inter-arrival times are exponentially distributed, is used to model the random arrival rates of the packets at each hop. An absorbing states Markov chain model was developed to determine the expected number of re-transmissions with CSMA/CA between each hop in a multi-hop scenario. The absorbing states are for when the seven retry limit is exceeded at each hop, and the packet gets discarded. The other absorbing state is for the destination node. Equations are derived to calculate the expected end-to-end delay by using the values obtained for the queue waiting time and the expected re-transmissions models. The total end-to-end delay is made up of the waiting times and system times. The waiting times at each hop node are made up of the access delay time, AIFS or DIFS and the back-off time which depends on the contention window (CW) size. The system time is made up of the time to transmit the packet, SIFS, time to transmit the ACK message, propagation delay, ACK-timeout period in the event that no ACK is received and the number of transmissions at each hop link. To use the Markov chain model developed, the probability of successful transmission on the medium is required. This probability varies depending on the network size, the load level in the network and the number of nodes in the interference range.

In order to evaluate the accuracy of the proposed analytical model, a comparison of the sim-



(a) 50 pps for High Priority Data - Scenario 3
(b) 50 pps for Medium Priority Data - Scenario 3
(c) 50 pps for Low Priority Data - Scenario 3
(d) 100 pps for High Priority Data - Scenario 4
(e) 100 pps for Medium Priority Data - Scenario 4
(f) 100 pps for Low Priority Data - Scenario 4
(g) 150 pps for High Priority Data - Scenario 5
(h) 150 pps for Medium Priority Data - Scenario 5
(i) 150 pps for Low Priority Data - Scenario 5

Figure 7.8: End-to-End Delay with the three queues RWS scheduling scenarios.

ulated and analytical model end-to-end delay results was carried out by varying traffic loads for the different priority data classes. The results show close correlation with an error percentage of less than 17% in the worst cases.

The model works for the basic channel access mechanism of CSMA/CA and not for RTS/CTS. The overheads due to other layers are not considered in the model. This model is designed for schedule before contention scheduling strategies in single-radio single-channel multi-hop networks. The model is designed to work when the system is stable with a utilization factor of below 1. The number of re-transmissions plays a critical role in end-to-end delay and reliability QoS achievable in multi-hop networks. The model also indicates the collision probability expected in a multi-hop network. This model is important in view of the rapid research and implementations being carried out to use CSMA/CA in wireless multi-hop networks for extending networks. Collisions waste bandwidth which is an important factor in determining the success of networks where bandwidth is limited.

Chapter 8

Summary and Conclusions

8.1 Summary

WMNs have been facing multiple limitations such as limited capacity due to the shared medium. In multi-hop networks, the contention increases for the medium as packets have to be transmitted in a multi-hop manner to reach the destination depending on the location of the destination. An increase in contention results in an increase in collisions. The situation is worsened in multi-hop networks in high load scenarios compared to single-hop networks. CSMA/CA and EDCA have been mainly developed for single-hop networks. EDCA is known to present unfairness and starvation problems for lower priority data as well as high collision rates for higher priority data in high load scenarios. EDCA has been mainly developed to provide QoS support at the MAC layer for multimedia data. The key challenges in application of carrier sense multiple access with collision avoidance (CSMA/CA) in multi-hop WMN networks based on single-radio single-channel networks is improving quality of service (QoS) by reducing collisions, reducing packet loss and improving intra-node fairness under heavy load scenarios. The focus of the dissertation was to address these limitations that affect the performance in WMNs.

Five MAC layer scheduling strategies to address these limitations especially under heavy loads were investigated based on a schedule-before-contention (SBC) packet scheduling. These SBC strategies have different packet selection mechanisms and they do not have an internal contention mechanism as in the EDCA standard. The internal contention mechanism contributes to the unfairness and starvation problem. The scheduling mechanisms were developed such that the higher priority data are given higher chances for transmission and to access the medium. An investigation on performance in both homogeneous configured network layouts and hybrid configured network layouts was carried out. In the homogeneous configured network layout schemes, all the nodes were assigned the same scheduling protocol, while in the hybrid configured network layout, the edge and core nodes were assigned different scheduling strategies. The performance of the scheduling strategies was compared through simulations with DCF in the IEEE 802.11 standard and with EDCA in the IEEE 802.11e standard for data differentiated services. The best performing scheduling strategy was implemented on the FIT IoT-Lab test-bed and the performance in terms of packet loss reduction verified. A novel analytical model for the end-to-end delay for schedule before contention strategies that follow Markovian theory was also developed.

8.2 Conclusion

The work as recorded in this dissertation has shown that with differentiated data in a multi-hop WMN, the DCF performs better than EDCA in-terms of less packet loss and higher fairness. DCF does not provide differentiated services and treats all data equally. DCF use larger CW range values which reduce the collision probability but increases the end-to-end delay compared to EDCA. With EDCA, the end-to-end delay is lower for high and medium priority data classes but the packet loss is very high for high and medium priority data compared to DCF.

This section of the conclusion is structured such that the hypothesis investigated is presented with the findings summarized subsequently:

Hypothesis 1: A replacement of this internal contention mechanism with a predefined deterministic weighted round robin scheduling queue selection mechanism can improve fairness and reduce packet loss in multi-hop networks.

Research question 1 in section 4.2 on whether replacing the internal contention mechanism in EDCA with a weighted round robin scheduling mechanism reduce packet loss was used to find the answer to this hypothesis. A weighted round robin SBC mechanism was developed and its performance investigated. A packet loss reduction of between 9.6% and 18.4% on average calculated with AWRR over EDCA under high loads has been observed. The weighted round robin mechanism cycles from the high priority queue to the lower priority queues and transmits more packets from the higher priority queue if it has data. The mechanism increased the overall lower priority data transmissions under heavy loads compared to EDCA which results in lower collisions. AWRR improves fairness to 0.983 under heavy loads in the grid topology from 0.9240 with EDCA. These results clearly support this hypothesis.

Hypothesis 2: Scheduling strategies that do not starve lower priority data but give higher priority data a higher chance to access the medium can reduce packet loss and the number of collisions in WMNs compared to the EDCA contention based strategy

In section 4.2, the analysis to (1) research question 2 on which SBC mechanism results in the lowest packet loss and end-to-end delay and (2) research question 3, on whether in terms of packet loss, a SBC strategy is better than having an internal contention mechanism such as in EDCA, were used to provide investigation to this hypothesis.

Novel MAC layer scheduling strategies which remove the internal collision mechanism in EDCA and perform scheduling of data according to novel scheduling mechanisms were proposed and implemented. These SBC mechanisms were the adaptive weighted round robin (AWRR), the roulette wheel sampling (RWS), the RWS-AGE, the congestion control and fairness scheduling (CCFS) and queue load control priority (QLCP). These different SBC mechanisms have shown different performance on the test networks as can be seen from the results in chapter 4 and 5. The CCFS and QLCP mechanisms experienced higher packet loss than the AWRR, RWS and RWS-Age mechanisms under heavy load scenarios. The CCFS strategy changes the number of slots assigned to the different queues when the load exceeds the threshold value in any queue in the mechanism. This ends up lowering the overall transmission probability of the lower priority data and therefore, results in starvation of the lower priority data. The QLCP mechanism experiences less packet loss than CCFS. QLCP is designed to transmit packets from the longer queues first in the order of their priority. Therefore, with QLCP, the lower priority data have a higher chance of accessing the medium under heavy loads. The AWRR, RWS and RWS-AGE scheduling mechanisms are adaptive and change the number of slots or weights for each priority class depending on which queues have data. On average over all the test topologies, the RWS-AGE mechanism experienced lower packet loss than the RWS and AWRR mechanisms. The random probability weight assigned selection mechanism with an age counter (RWS-AGE) performs better than the mechanism without an age counter (RWS). RWS-AGE also performs better than a weighted round robin wheel for transmission of heterogeneous data. Although a reduction in packet loss compared to EDCA and DCF is observed with AWRR, RWS and RWS-AGE, the end-to-end delay is increased for high and medium priority data compared to EDCA. However, it is lower than DCF.

Research question 4 in section 4.2 on whether the use of TXOP in SBC strategies improve performance in terms of lowering packet loss and end-to-end delay was used to investigate if the operation of SBC strategies can be improved with the used of TXOP. The use of TXOP in the SBC mechanisms has been investigated and a significant packet loss reduction and end-to-end delay is observed. These results clearly support this hypothesis. The operation of the scheduling strategy has a significant effect on the achievable QoS in SBC strategies.

The use of a SBC with a scheduling mechanism that does not starve lower priority data under heavy loads is the key to reducing packet loss in multi-hop networks as shown in this work. The transmission of lower priority data plays a key role in reducing collisions as well as lowering packet loss in the network.

Hypothesis 3: A load control scheduling strategy in a hybrid configured network layout where different nodes are assigned different scheduling strategies with some of these devices assigned DCF can result in a reduction in packet loss over homogeneous configured EDCA network implementations.

A novel scheduling strategy for queue load control and fairness improvement called queue load control priority (QLCP) scheduling strategy for nodes that are subjected to higher load levels in a network was developed. In section 5.2, the analysis to (1) research question 2 whether QLCP perform better in homogeneous configured network layouts or hybrid configured network layouts in terms of packet loss and end-to-end delay and (2) research question 3 whether the use of DCF in hybrid configured network layouts improve performance, were used to provide investigation to this hypothesis. The performance of this strategy was analysed in both homogeneous configured network layouts (all nodes assigned the same scheduling strategy) and in hybrid configured network layouts (assigned different scheduling strategies) for networks with gateway nodes that are subject to high traffic loads. In the homogeneous configured network layout implementations as can be seen from chapter 5, QLCP lowers packet loss compared to EDCA. QLCP in the configured network layouts with QLCP configured in the edge nodes and DCF in the core nodes further reduces packet loss for high and medium priority data but increases the end-to-end delay. QLCP experiences 29.8% packet loss on average data in the DCF (C) and QLCP (E) settings compared to 41.6% packet loss on average in its homogeneous configured network layout implementation under heavy loads. QLCP performs better in the hybrid setting with QLCP configured in the edge nodes and DCF in the core nodes. The reduction in packet loss for the hybrid configured network layout with the core nodes assigned DCF supports the hypothesis.

Hypothesis 4: In networks with differentiable edge and core nodes, hybrid configured network layouts with the use of DCF can reduce packet loss as well as the number of collisions compared to their homogeneous configured network layout implementation.

Research question 1 whether RWS-AGE perform better in homogeneous configured network layouts or hybrid configured network layouts in terms of packet loss and end-to-end delay and research question 4 on which overall design combination gives the best performance in multi-domain networks in section 5.2 were used to find answers to this hypothesis. Hybrid configured network layouts with DCF improved performance over their homogeneous configured network layout implementations, as can be seen from the results in chapter 5. The different priority traffic carried in the backbone nodes configured with DCF gain access to the medium in a FIFO manner with same channel access parameter values. The edge nodes configured with EDCA or QLCP give a higher chance to the higher priority data to access the network. The layout where EDCA was configured in the edge nodes and DCF in the core nodes experienced the best performance in terms of less packet loss and collisions; and least end-to-end delay. In these hybrid configured network layouts, the core nodes are basically performing non-differentiated data services on a FIFO basis and the edge nodes are performing data differentiated services according to the data priority. On the other hand, the hybrid networks that used DCF in the edge nodes showed to lower packet loss, reduce collisions but increase end-to-end delay. The hybrid layout with DCF configured in the edge nodes and RWS configured in the core nodes showed the least packet loss and low end-to-end delay. This clearly supports the hypothesis and shows that hybrid configured network layouts with the use of DCF, reduce packet loss.

The work of this dissertation has shown that removing the internal collision mechanism in EDCA and using a scheduling mechanism that does not starve lower priority data in single radio single channel networks is an important ingredient to reducing packet loss, reducing collisions and improving fairness in multi-hop WMNs. The investigations have also shown that the network layout and scheduling strategies used, play a major role on the achievable QoS.

The roulette wheel scheduling strategy with an age counter was implemented in the Contiki operating system on sensor nodes on the FIT-IoT Lab test bed and packet loss reduction performance improvement was observed at heavy loads. Contiki is open source and allows us to use, modify and make additions to this operation system. It was used to develop an analytical model to calculate the end-to-end delay for the different priority data classes for the RWS and DCF scheduling strategies that follow the schedule before contention principle. The model was developed for the basic CSMA/CA mechanism and holds under stable network utilization conditions. This model is applicable to both networks with nodes with a single queue or multiple queues for differentiated heterogeneous data. The analytical model applied Markov chain theory with absorbing states and queuing theory. The model consists of two sub-models, namely one to calculate the access waiting queuing time through application of traffic theory and the other is an absorbing Markov chain model to determine the number of re-transmissions at each hop in a multi-hop network. The values from these sub-models are then integrated with the other channel transmission parameters such as AIFS, back-off period, packet transmission time, propagation delay, ACK time and retransmission times to approximate the end-to-end delay. To use the Markov chain model developed, the probability of successful transmission on the medium is required, which depends on the network size, the CW sizes and the load level in the network. The accuracy of the model has been verified through compared simulation and analytical results for multi-hop networks. The use of this model is only applicable to schedule before contention strategies that obey the Markovian propriety. CCFS and AWRR do not obey the Markovain property, as their operation depends on history such as the Age counter, as well as the number of packets from each that have been served.

8.3 Guidelines

The results from the investigations presented support to the initial hypotheses and provide important guidelines for the implementation and use of the proposed scheduling strategies in SRSC WMNs. The network layout and design plays a critical role in determining the choice of scheduling strategies and QoS achievable. The choice of the scheduling strategy based on the network design is an important ingredient in coordinating access to the medium in an effective manner to achieve efficient QoS in SRSC WMNs. The guidelines below can be used to select scheduling strategies for multi-hop networks:

1. For networks with different domains connecting to the backbone mesh network through edge nodes:
    - For networks with gateway nodes and that require high reliability, but can tolerate slightly more end-to-end delay, a hybrid configured network layout, where DCF is configured in the edge nodes and RWS-AGE is configured in edge nodes, will be a good design.
    - For networks that require low end-to-end delay and can tolerate slight packet loss, a hybrid configured network layout, where DCF is configured in the core nodes and EDCA is configured in edge nodes will be a good design choice.
2. Networks where the edge and core nodes are not differentiable:
    - The RWS-AGE scheduling strategy will be a good choice in terms of least packet loss, high fairness and low collisions over RWS, AWRR, CCFS, DCF and EDCA scheduling strategies.

8.4 Summary of Contributions

The research as documented herein made the following contributions:

1. Five MAC layer schedule-before-contention (SBC) packet scheduling strategies to improve fairness and reduce collisions have been developed. These are the AWRR, RWS, RWS-AGE, CCFS and QLCP scheduling strategies. All these strategies have different packet selection mechanisms based on the SBC concept. The performance of these strategies was compared with EDCA and DCF in multi-hop network settings. The following findings have been observed:
    - The DCF performs better in multi-hop networks than EDCA in terms of less packet loss.
    - The removal of the internal contention mechanism in EDCA with a scheduling mechanism that do not starve lower priority data such as AWRR, RWS and RWS-AGE does improve fairness under heavy loads, as well as reduce packet loss.
    - The QLCP scheduling strategy (developed for nodes that are subjected to higher load levels in a network) performs well in hybrid configured network layouts with the edge routers configured with QLCP and the core routers configured with DCF.
    - The RWS-AGE strategy is ideal for implementation in homogeneous configured network layouts with all the nodes assigned the same scheduling strategy for networks that require high reliability and can tolerate slightly higher delay than EDCA.

- In hybrid configured network layouts where it is possible to differentiate between edge and core nodes, the use of DCF has shown to improves performance.
- It has been shown that the choice of the scheduling strategy must be dependent on the network architecture and has shown to have a significant impact on the QoS achievable.
- The use of TXOP in SBC strategies has shown to reduce packet loss and end-to-end delay.

2. The RWS-AGE strategy was implemented in the Contiki operating system on the FIT-IoT Lab test bed and packet loss reduction performance improvement observed over the default CSMA/CA mechanism.

3. An analytical model for the end-to-end delay for schedule-before-contention has been developed and tested with the RWS and DCF strategies. The analytical model consists of the access delay model, an absorbing state Markov chain model to determine the expected number of transmissions and derived equations to calculate the expected end-to-end delay by using the values obtained from the access delay model and the expected transmission model. There is a good correlation between the model and simulation results.

4. Guidelines based on the findings on the performance of these scheduling strategies have been proposed for network planning, implementation and network optimization in multi-hop networks.

8.5 Future Research Directions

Based on the work as presented in this dissertation, this section presents areas for possible future research:

1. Throughput capacity achievable for WMNs nodes is limited in single-channel systems compared to multi-channel systems. The application of the scheduling strategies in multi-channel systems can be investigated.

2. In our survey paper in [84], some challenges in WMNs that have not been completely solved have been presented. This work addressed the distributed priority fair scheduling problem. Other challenges include scalability and throughput decrease in multicast applications in WMNs.

3. In our survey paper in [84], we have argued that link quality for routing in WMNs also plays a significant role towards the QoS. Joint Scheduling and routing techniques based on the scheduling strategies as developed in this work can be expanded and investigated for further optimization.

4. The mathematical model as proposed does not provide for packet loss. The model can be extended to include packet loss by incorporating details of other layers, such as the routing protocol and physical layer.

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Appendices

Publications and Author's Contributions

At the start of the PhD work, the main objective was to contribute to the enhancement of QoS in terms of improving reliability for heterogeneous data in multiple-hop resource constrained WMNs for wireless telemetry and smart applications. The main aim is to use lower cost communication technology for cost effective wireless communication. Some of the work in this dissertation has been published in the following publications. In paper 1, a performance analysis for hybrid and pure design layouts with EDCA and DCF in multi-hop WMNs for heterogeneous data is carried out. The concept of assigning different scheduling strategies to different devices is novel. A book chapter was written on scheduling strategies to improve reliability and fairness for priority based smart rural contention based applications. The chapter carries out further tests to support the idea of hybrid design layouts presented in paper 1. In paper 2, a novel AWRR strategy is proposed and the performance of this strategy is compared to EDCA. The proposed strategy is a schedule-before-contention (SBC) strategy. In paper 3, a novel adaptive CCFS strategy which is also a SBC strategy is proposed and its performance compared to EDCA. In paper 4, a survey of cross-layer approaches using the IEEE 802.11 standard in WMNs has been carried out. This paper also presents open issues that need research attention. Paper 5, carries out an investigation on the impact of the transmission opportunity (TXOP) on the performance of priority based contention based scheduling strategies in multi-hop mesh networks to provide further optimization. In paper 6, a novel random priority based SBC strategy is proposed and its performance tested over a testbed of wireless sensor networks. In paper 7, the hybrid and pure design concept if further tested with novel SBC strategies namely RWS and QLCP for reliability and load control in multi-domain low cost WMNs for heterogeneous data. In paper 8, modeling of the proposed Schedule-before-Contention MAC Strategies in wireless multi-hop networks is carried out. In paper 9, a comparative analysis of MANET routing protocols for low cost rural telemetry WMNs was carried out.

Paper 1: Performance and Comparative Analysis of Design Schemes for Prioritised Data in Multi-Hop Wireless Mesh Backbone Networks

Abstract: The contention based carrier sense multiple access with collision avoidance (CS-MA/CA) was originally designed for single-hop networks. For CSMA/CA to be used in multi-hop distributed networks and to provide guaranteed data priority, the CSMA/CA needs to be optimised. An application is the smart grid consisting of different network domains with data of different priority levels. The IEEE 802.11e standard was developed to provide differentiated data services. With the default enhanced distributed channel access (EDCA) settings for QoS,

an unfairness problem exists for different data classes where higher priority data can starve lower priority data and also where bandwidth is allocated unfairly. In this paper, we carry out an investigation of six design schemes for wireless backbone networks for QoS provisioning of different data priority classes. The design schemes are based on the concept of low-cost design for suitability in rural areas where cost plays a major role. The simulation results were obtained using OMNeT++ and the INET framework. The performance metrics used for the analysis were end-to-end latency, packet loss percentage and Jain's fairness index. Simulation results show that hybrid network designs using distributed coordination function (DCF) and EDCA can improve QoS in terms of reliability and fairness.

Lessons Learnt: The hybrid design scheme where DCF was configured in the core routers and EDCA in the edge routers experienced the least packet loss. This was due to a reduction in the number of collisions as DCF have larger CW ranges and contention periods compared to the differentiated IEEE 802.11e services differentiation scheme.

Published in: Proceedings of the International Conference on Wireless Information Networks and Systems (WINSYS 2015), 20-22 July 2015 held in Colmar, Alsace, France.

Achievements: The paper was awarded the Best Student Paper Award Certificate and a further invitation was received to write a book chapter.

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Book Chapter: Scheduling Strategies to Improve Reliability and Fairness for Priority Based Smart Rural Contention Based Applications over Low-Cost Wireless Mesh Backbone Networks

Abstract: Wireless Mesh Networks (WMNs) are viewed as a cheap solution for telemetry networks in rural areas. The main advantages of WMNs are that they allow an easy extension of existing networks to service a wider area by using multi-hop wireless communication and they provide an alternate route when a route becomes faulty. Smart Rural Areas is a new concept for the development of rural areas. It is hypothesized that the Internet of Things (IoT) can help develop rural areas by providing better services resulting in poverty reduction. The widely used carrier sense multiple access with collision avoidance (CSMA/CA) was originally designed for Wireless Local Area Networks (WLANs) consisting of single-hop transmissions. CSMA/CA experiences a rapid decrease in performance when applied to multi-hop distributed networks as an increase in collisions and contention for the medium is experienced. The IEEE 802.11e standard provides data differentiation services for data of different priority levels with enhanced distributed channel access (EDCA) being used in contention based networks. With EDCA, an unfairness problem exists where high priority data can starve lower priority data. To address these problems in low-cost rural smart networks we investigate the performance of six design schemes for wireless backbone networks by assigning different roles to edge and core routers. Simulations were carried out to obtain the results using OMNeT++ and the INET framework. Simulation results show that hybrid network designs using distributed coordination function (DCF) and EDCA can improve QoS.

Lessons Learnt: It was realised that the pure (homogeneous configured network layout) DCF design schemes with all routers configured with DCF experienced less packet loss, but high end-to-end delay. The pure EDCA design schemes with all devices configured with EDCA, experienced the highest packet loss and the lowest end-to-delay for medium and low priority data, but the highest end to end-to-end delay for low priority data. The hybrid design scheme where DCF was configured in the core routers and EDCA in the edge routers experienced the lowest end-to-end delay and low packet loss compared with the other hybrid design configured with DCF in the edge routers and EDCA in the core which experienced lower packet loss but higher end-to-end delay. The scheduling strategy assignment to different nodes in the network plays a critical role on the performance. Hybrid schemes can improve performance.

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Paper 2: A Cross Layer Adaptive Weighted Round Robin Scheduling Strategy for Wireless Mesh Networks

Abstract: Over the last few years, Wireless Mesh Networks (WMNs) have been experiencing an increase in deployment and research by both the business community and academia. In WMNs, the time division multiple access (TDMA) and code division multiple access (CDMA) medium access control (MAC) contention-free protocols require time synchronisation in the global network. This is difficult to achieve in multi-hop networks. The distributed carrier sense multiple access with collision avoidance (CSMA/CA) contention based protocol is more suitable in WMNs. In many real world applications, higher priority data need to be delivered with guaranteed Quality of Service (QoS). The IEEE 802.11e standard was developed for differentiated services. The enhanced distributed channel access (EDCA) used with CSMA/CA experiences unfairness problems where higher priority data can starve lower priority data and also where bandwidth is allocated unfairly. In this paper, we propose a novel cross-layer scheduling technique called adaptive weighted round robin (AWRR) to address the fairness and reliability problems in WMNs using CSMA/CA. Simulations are carried out in OMNeT++ using the INETMANET framework to ascertain the performance of the proposed strategy. The performance metrics used for the analysis and study are end-to-end latency, packet loss percentage and Jain's fairness index. The proposed strategy is shown to reduce packet loss of up to 30%.

Lessons Learnt: A reduction in packet loss is observed as there is no internal collisions are encountered due to the deterministic scheduling performed. The end-to-end delay is increased.

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Paper 3: An Adaptive Congestion Control and Fairness Scheduling Strategy for Wireless Mesh Networks

Abstract: Wireless mesh networks (WMNs) are a promising technology for low cost deployments for telemetry networks in rural areas. The popular contention based carrier sense multiple access with collision avoidance (CSMA/CA) technique is widely used in WMN implementations as it does not require time synchronization compared to time division multiple access (TDMA). The IEEE 802.11e standard was introduced to provide data differentiation services to data on a network with data of different priority. With this standard, the enhanced distributed channel access (EDCA) technique for contention based services experiences a fairness problem where high data can starve lower priority data. CSMA/CA was originally developed for single-hop networks. Collisions tend to increase in multi-hop networks as the contention for the medium increases. To address the fairness and performance degradation with an increase in contention in multi-hop network problems, a novel adaptive congestion control and fairness scheduling (CCFS) strategy is proposed in this paper. The proposed strategy is simulated in OMNeT++ using the INETMANET library to ascertain the performance of the strategy. The strategy was compared with EDCA in terms of end-to-end latency, packet loss percentage and Jain's fairness index. The proposed adaptive strategy is shown to reduce packet loss in most test cases as well as provide an overall more fair system with data of different priority when compared to EDCA.

Lessons Learnt: Simulation results show a reduction in packet loss in most cases with this strategy but an increase in end-to-end latency for high and medium priority data compared to EDCA. The strategy also discards the internal collision mechanism present in EDCA.

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Paper 4: A Survey of Cross-Layer Protocols for IEEE802.11 Wireless Mesh Networks

Abstract: There has been an escalation in deployment and research of Wireless Mesh Networks (WMNs) by both the business community and academia in the last few years. Their attractive characteristics include low deployment cost, a low-cost option to extend network coverage, and ease of maintenance due to their self-healing properties. Multiple routes exist between the sender and receiver nodes due to the mesh layout which ensure network connectivity even when node or link failures occur. Recent advances among others include routing metrics, optimum routing, security, scheduling, cross-layer designs and physical layer techniques. However, there are still challenges in WMNs as discussed in this paper that need to be addressed. Cross-layer design allows information from adjacent and non-adjacent layers to be used at a particular layer for performance improvement. This paper presents a survey of cross-layer protocol design approaches applied to the IEEE 802.11 standards for wireless multi-hop mesh networks that have been proposed over the last few years for improved performance. We summarise the current research efforts in cross-layer protocol design using the IEEE 802.11 standard in identifying unsolved

issues which are a promising avenue to further research.

Lessons Learnt: Four main areas of unsolved issues in WMNs are scalability; application specific routing; unfairness of prioritised data; and multicast routing and multi-rate scheduling strategies to save bandwidth in multicast applications have been identified in this survey. Limited work has been done for resource constraint networks.

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Paper 5: The Impact of Transmission Opportunity (TXOP) on the Performance of Priority Based Contention Based Scheduling Strategies in Multi-hop Mesh Networks

Abstract: Wireless Mesh Networks (WMNs) face multiple problems. An increase in the number of hops for packets to reach the destination results in an increase in contention for the medium which also results in an increase in the collision rates. The enhanced distributed channel access (EDCA) mechanism was developed to provide differentiated services to data with different priority levels in the IEEE 802.11e standard. The EDCA is a distributed, contention-based channel access mechanism of the hybrid coordination function (HCF) which results in an unfairness problem where higher priority data can starve lower priority data. We adopt the EDCA architecture for heterogeneous data in telemetry and IoT applications to address these problems of EDCA in multi-hop mesh networks. An adaptive weighted round robin (AWRR) scheduling strategy has been proposed and tested on multi-hop networks in our previous work. With the AWRR strategy, although packet loss is reduced, the end-to-end delay increases with high and medium priority data compared to EDCA in WMNs. In this paper we investigate the effect of the Transmission Opportunity (TXOP) bursting on the global performance in a WMN through setting up simulations in OMNeT++ using the INETMANET framework. Simulation results have shown that using TXOP bursting in the priority based scheduling which follows the concept of schedule before backup helps reduce packet loss as well as reduce the end-to-end delay. TXOP further optimizes the performance of AWRR.

Lessons Learnt: The advantage of TXOP is that multiple packets from the same queue can be transmitted without the need of continuously performing the contention period. With AWRR using TXOP, a reduction in collisions has been shown as the channel is sensed as being busy by the other nodes during the TXOP period and the other packets within the TXOP period of the same data class can successful transmit. Re-transmission of collided packets waste channel bandwidth and reduce the overall performance of the network. This paper has shown that the performance of AWRR can be further improved and optimised by the use of TXOP limit values by reducing end-to-end delay and reducing packet loss.

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Paper 6: A Random Priority Based Scheduling Strategy for Wireless Sensor Networks Using Contiki

Abstract: In recent years, wireless sensor networks (WSNs) have experienced a number of implementations in various implementations which include smart home networks, smart grids, smart medical monitoring, telemetry networks and many more. The Contiki operating system for wireless sensor networks which utilises carrier sense multiple access with collision avoidance (CSMA/CA) does not provide differentiated services to data of different priorities and treats all data with equal priority. Many sensor nodes in a network are responsible not only for sending their sensed data, but also forwarding data from other nodes to the destination. In this paper we propose a novel priority data differentiation medium access control (MAC) strategy to provide differentiated services called Random Weighted Scheduling (RWS). The strategy was implemented and tested on the FIT IoT-lab testbed. The strategy shows a reduction in packet loss compared to the default CSMA/CA scheduling strategy in IEEE 802.15.4 WSNs when carrying data of different priority levels.

Lessons Learnt: The RWS scheduling strategy was developed and implemented in the Contiki operating system which is an open source. The Rime protocol communication stack was used as the other layers are light weight and this helps to ascertain the performance of the proposed scheme. RWS has shown a reduction in packet loss as the number of hops is increased for most of the test cases implemented over the FIT IoT-lab test bed.

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Paper 7: A Model Analyzing the Performance of Analysis of Wireless Multi-hop Networks using a Contention-based CSMA/CA Strategy

Abstract: Multi-hop networks are a low-setup-cost solution for enlarging an area of network coverage through multi-hop routing. Carrier sense multiple access with collision avoidance (CSMA/CA) is frequently used in multi-hop networks. Multi-hop networks face multiple problems, such as a rise in contention for the medium, and packet loss under heavy-load, saturated conditions, which consumes more bandwidth due to re-transmissions. The number of re-transmissions carried out in a multi-hop network plays a major role in the achievable quality of service (QoS). This paper presents a statistical, analytical model for the end-to-end delay of contention-based medium of access control (MAC) strategies. These strategies schedule a packet before performing the back-off contention for both differentiated heterogeneous data and homogeneous data under saturation conditions. The analytical model is an application of Markov chain theory and queuing theory. The M/M/1 model is used to derive access queue waiting times, and an absorbing Markov chain is used to determine the expected number of re-transmissions in a multi-hop scenario. This is then used to calculate the expected end-to-end delay. The prediction by the

proposed model is compared to the simulation results, and shows close correlation for the different test cases with different arrival rates.

Lessons Learnt: The number of re-transmissions plays a critical role in end-to-end delay and reliability QoS achievable in multi-hop networks. The model indicates the collision probability expected in a multi-hop network as well as the number of re-transmissions. Collisions waste bandwidth which is an important factor in determining the success of networks where bandwidth is limited.

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Other

Some of the work initially done as part of the PhD, but is not in the scope of this dissertation is as follows:

Paper 8: A Comparative Analysis of MANET Routing Protocols for Low Cost Rural Telemetry Wireless Mesh Networks

Abstract: In rural areas in Africa, the topographical conditions vary, including hilly areas or flat open areas with bushes, trees and vegetation. In some cases, road and infrastructure conditions are exceedingly poor, making it challenging and costly to provide necessary maintenance and support to communication networks. When a node goes offline the remaining nodes must be able to re-establish links with each other and maintain connectivity. The routing protocol must discover an alternative shortest path route and use this path to deliver the data. The maintenance time can be slow and it might take days to attend to the faulty node in a rural area. Due to this, the network must be able to operate for long periods with the faulty node(s) and provide the best possible Quality of Service (QoS). In the past few years, Wireless Mesh Networks (WMNs) have attracted an increase in research and use due to their attractive characteristics, which include low deployment cost, a low cost option to extend network coverage and ease of maintenance due to their self healing properties. In WMNs, with an increase in scalability, the throughput of the network tends to decrease. In this paper, we carried out a performance analysis for failing node scenarios for rural telemetry networks using three protocols, namely OLSR (a proactive protocol), DSR (a reactive protocol) and HWMP (a hybrid protocol). The performance analysis of these protocols was carried out using three backhaul network topology scenarios. The simulation results were obtained using OMNeT++ and the INETMANET framework. Performance metrics used for the analysis and study were packet loss and end-to-end latency as these are major factors considered for providing guaranteed Quality of Service (QoS).

Lessons Learnt: HWMP protocol had the least packet loss compared to DSR and OLSR for networks with frequently changing topologies. The OLSR protocol was optimized by reducing the hello messages interval and by reducing the topology control messages emission interval. Although the performance of OLSR was improved considerably, the performance was still lower with HWMP in terms of packet loss. For a rural telemetry network using WMNs under conditions of changing network topologies, the HWMP will be a better choice due to its better reliability in terms of less packet loss and low end-to-end latency.

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